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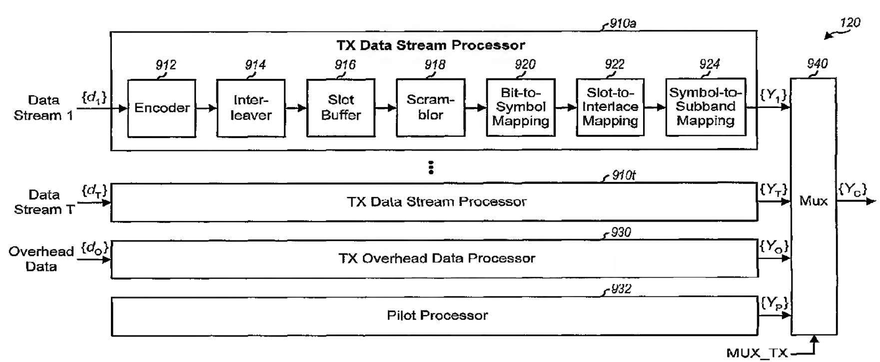
(54) Title: FREQUENCY DIVISION MULTIPLEXING OF MULTIPLE DATA STREAMS IN A WIRELESS MULTI-CARRIER COMMUNICATION SYSTEM



(57) Abstract: Techniques for multiplexing multiple data streams using frequency division multiplexing (FDM) in an OFDM system are described. M disjoint "interlaces" are formed with U usable subbands. Each interlace is a different set of S subbands, where . The subbands for each interlace are interlaced with the subbands for each of the other interlaces. M slots may be defined for each symbol period and assigned slot indices 1 through M. The slot indices may be mapped to interlaces such that (1) frequency diversity is achieved for each slot index and (2) the interlaces used for pilot transmission have varying distances to the interlaces used for each slot index, which improves channel estimation performance. Each data stream may be processed as data packets of a fixed size, and different numbers of slots may be used for each data packet depending on the coding and modulation scheme used for the data packet.

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FREQUENCY DIVISION MULTIPLEXING OF MULTIPLE DATA STREAMS IN A WIRELESS MULTI-CARRIER COMMUNICATION SYSTEM

[0001] This application claims the benefit of U.S. Application Serial No. 10/932,586, entitled "A Method for Multiplexing and Transmitting Multiple Multimedia Streams to Mobile Terminals over Terrestrial Radio," filed September 1, 2004, provisional U.S. Application Serial No. 60/559,740, entitled "Multiplexing and Transmission of Multiple Data Streams in a Wireless Multi-Carrier Communication System," filed April 5th, 2004, and provisional U.S. Application Serial No. 60/514,315, entitled "A Method for Frequency-Division Multiplex Various Multimedia Streams for Multicast Wireless Transmission to Mobile Devices," filed October 24, 2003.

BACKGROUND

I. Field

[0002] The present invention relates generally to communication, and more specifically to techniques for multiplexing multiple data steams in a wireless multi-carrier communication system.

II. Background

[0003] A multi-carrier communication system utilizes multiple carriers for data transmission. These multiple carriers may be provided by orthogonal frequency division multiplexing (OFDM), some other multi-carrier modulation techniques, or some other construct. OFDM effectively partitions the overall system bandwidth into multiple (N) orthogonal frequency subbands. These subbands are also referred to as tones, carriers, subcarriers, bins, and frequency channels. With OFDM, each subband is associated with a respective subcarrier that may be modulated with data.

[0004] A base station in a multi-carrier communication system may simultaneously transmit multiple data streams. Each data stream may be processed (e.g., coded and modulated) separately at the base station and may thus be recovered (e.g., demodulated and decoded) independently by a wireless device. The multiple data streams may have fixed or variables data rates and may use the same or different coding and modulation schemes.

[0005] Multiplexing multiple data streams for simultaneous transmission may be challenging if these streams are variable in nature (e.g., have data rates and/or coding and modulation schemes that change over time). In one simple multiplexing scheme, the multiple data streams are allocated different time slots or symbol periods using time division multiplexing (TDM). For this TDM scheme, only one data stream is sent at any given moment, and this data stream uses all subbands available for data transmission. This TDM scheme has certain undesirable characteristics. First, the amount of data that may be sent in the smallest time unit allocable to a given data stream, which may be viewed as the "granularity" for the data stream, is dependent on the coding and modulation scheme used for the data stream. Different coding and modulation schemes may then be associated with different granularities, which may complicate the allocation of resources to the data streams and may result in inefficient resource utilization. Second, if the granularity for a given coding and modulation scheme is too large relative to the decoding capability of a wireless device, then a large input buffer may be required at the wireless device to store received symbols.

[0006] There is therefore a need in the art for techniques to efficiently multiplex multiple data streams in a multi-carrier communication system.

SUMMARY

[0007] Techniques for multiplexing multiple data streams using frequency division multiplexing (FDM) in a wireless multi-carrier (e.g., OFDM) communication system are described herein. In an embodiment, M disjoint or non-overlapping "interlaces" are formed with U subbands usable for transmission, where $M > 1$ and $U > 1$. The interlaces are non-overlapping in that each usable subband is included in only one interlace. Each interlace is a different set of S subbands, where $U = M \cdot S$. The S subbands in each interlace may be selected from S' subbands that are uniformly distributed across the N total subbands and are evenly spaced apart by M subbands, where $N = M \cdot S'$ and $S' \geq S$. This interlaced subband structure can provide frequency diversity and simplify processing at a receiver. For example, the receiver may perform a "partial" S'-point fast Fourier transform (FFT) for each interlace of interest, instead of a full N-point FFT. The M interlaces may be used to transmit the multiple data streams in an FDM manner. In an embodiment, each interlace is used by only one data stream

in each symbol period, and up to M data streams may be sent on the M interlaces in each symbol period.

[0008] In an embodiment, the multiple data streams are allocated "slots", where each slot is a unit of transmission that may be equal to one interlace in one symbol period. M slots are then available in each symbol period and may be assigned slot indices 1 through M. Each slot index may be mapped to one interlace in each symbol period based on a slot-to-interlace mapping scheme. One or more slot indices may be used for an FDM pilot, and the remaining slot indices may be used for data transmission. The slot-to-interlace mapping may be such that the interlaces used for pilot transmission have varying distances to the interlaces used for each slot index in different OFDM symbol periods. This allows all slot indices used for data transmission to achieve similar channel estimation performance.

[0009] Each data stream may be processed as data packets of a fixed size. In this case, different numbers of slots may be used for each data packet depending on the coding and modulation scheme used for the data packet. Alternatively, each data stream may be processed as data packets of variable sizes. For example, the packet sizes may be selected such that an integer number of data packets is sent in each slot. In any case, if multiple data packets are sent in a given slot, then the data symbols for each data packet may be distributed across all subbands used for the slot, so that frequency diversity is achieved for each data packet sent in the slot.

[0010] Various aspects and embodiments of the invention are described in further detail below.

BRIEF DESCRIPTION OF THE DRAWINGS

[0011] The features and nature of the present invention will become more apparent from the detailed description set forth below when taken in conjunction with the drawings in which like reference characters identify correspondingly throughout and wherein:

[0012] FIG. 1 shows a block diagram of a base station and a wireless device;

[0013] FIG. 2 shows an exemplary super-frame structure;

[0014] FIG. 3 shows an interlaced subband structure;

[0015] FIGS. 4A and 4B show "staggered" and "cycled" FDM pilots, respectively;

[0016] FIG. 5 shows an exemplary mapping of slot indices to interlaces;

[0017] FIG. 6 illustrates coding of a data block with an outer code;

[0018] FIGS. 7A and 7B show transmission of packets for different modes;

[0019] FIGS. 8A and 8B show partitioning of different numbers of packets into slots;

[0020] FIG. 9A shows a block diagram of a transmit (TX) data processor;

[0021] FIG. 9B shows a block diagram of a modulator;

[0022] FIG. 10A shows a block diagram of a demodulator; and

[0023] FIG. 10B shows a block diagram of a receive (RX) data processor.

DETAILED DESCRIPTION

[0024] The word "exemplary" is used herein to mean "serving as an example, instance, or illustration." Any embodiment or design described herein as "exemplary" is not necessarily to be construed as preferred or advantageous over other embodiments or designs.

[0025] The multiplexing techniques described herein may be used for various wireless multi-carrier communication systems. These techniques may also be used for the downlink as well as the uplink. The downlink (or forward link) refers to the communication link from the base stations to the wireless devices, and the uplink (or reverse link) refers to the communication link from the wireless devices to the base stations. For clarity, these techniques are described below for the downlink in an OFDM-based system.

[0026] **FIG. 1** shows a block diagram of a base station 110 and a wireless device 150 in a wireless system 100 that utilizes OFDM. Base station 110 is generally a fixed station and may also be referred to as a base transceiver system (BTS), an access point, a transmitter, or some other terminology. Wireless device 150 may be fixed or mobile and may also be referred to as a user terminal, a mobile station, a receiver, or some other terminology. Wireless device 150 may also be a portable unit such as a cellular phone, a handheld device, a wireless module, a personal digital assistant (PDA), and so on.

[0027] At base station 110, a TX data processor 120 receives multiple (T) data streams (or "traffic" data) and processes (e.g., encodes, interleaves, and symbol maps) each data stream to generate data symbols. As used herein, a "data symbol" is a modulation symbol for traffic data, a "pilot symbol" is a modulation symbol for pilot (which is data that is known *a priori* by both the base station and wireless devices), and a modulation symbol is a complex value for a point in a signal constellation for a

modulation scheme (e.g., M-PSK, M-QAM, and so on). TX data processor 120 also multiplexes the data symbols for the T data streams and pilot symbols onto the proper subbands and provides a composite symbol stream. A modulator 130 performs OFDM modulation on the multiplexed symbols in the composite symbol stream to generate OFDM symbols. A transmitter unit (TMTR) 132 converts the OFDM symbols into analog signals and further conditions (e.g., amplifies, filters, and frequency upconverts) the analog signals to generate a modulated signal. Base station 110 then transmits the modulated signal from an antenna 134 to wireless devices in the system.

[0028] At wireless device 150, the transmitted signal from base station 110 is received by an antenna 152 and provided to a receiver unit (RCVR) 154. Receiver unit 154 conditions (e.g., filters, amplifies, and frequency downconverts) the received signal and digitizes the conditioned signal to generate a stream of input samples. A demodulator 160 performs OFDM demodulation on the input samples to obtain received symbols for one or more data streams of interest, and further performs detection (e.g., equalization or matched filtering) on the received symbols to obtain detected data symbols, which are estimates of the data symbols sent by base station 110. An RX data processor 170 then processes (e.g., symbol demaps, deinterleaves, and decodes) the detected data symbols for each selected data stream and provides decoded data for that stream. The processing by demodulator 160 and RX data processor 170 is complementary to the processing by modulator 130 and TX data processor 120, respectively, at base station 110.

[0029] Controllers 140 and 180 direct operation at base station 110 and wireless device 150, respectively. Memory units 142 and 182 provide storage for program codes and data used by controllers 140 and 180, respectively. Controller 140 or a scheduler 144 may allocate system resources for the T data streams.

[0030] Base station 110 may transmit the T data streams for various services such as broadcast, multicast, and/or unicast services. A broadcast transmission is sent to all wireless devices within a designated coverage area, a multicast transmission is sent to a group of wireless devices, and a unicast transmission is sent to a specific wireless device. For example, base station 110 may broadcast a number of data streams for multimedia (e.g., television) programs and for multimedia content such as video, audio, teletext, data, video/audio clips, and so on. A single multimedia program may be broadcast as three separate data streams for video, audio, and data. This allows for

independent reception of the video, audio, and data portions of the multimedia program by a wireless device.

[0031] **FIG. 2** shows an exemplary super-frame structure 200 that may be used for system 100. The T data streams may be transmitted in super-frames, with each super-frame having a predetermined time duration. A super-frame may also be referred to as a frame, a time slot, or some other terminology. For the embodiment shown in FIG. 2, each super-frame includes a field 212 for one or more TDM pilots, a field 214 for overhead/control data, and a field 216 for traffic data. The TDM pilot(s) may be used by a wireless device for synchronization (e.g., frame detection, frequency error estimation, timing acquisition, and so on). The overhead/control data may indicate various parameters for the T data streams (e.g., the coding and modulation scheme used for each data stream, the specific location of each data stream within the super-frame, and so on). The T data streams are sent in field 216. Although not shown in FIG. 2, each super-frame may be divided into multiple (e.g., four) equal-sized frames to facilitate data transmission. Other frame structures may also be used for system 100.

[0032] **FIG. 3** shows an interlaced subband structure 300 that may be used for system 100. System 100 utilizes an OFDM structure having N total subbands. U subbands may be used for data and pilot transmission and are called "usable" subbands, where $U \le N$. The remaining G subbands are not used and are called "guard" subbands, where $N = U + G$. As an example, system 100 may utilize an OFDM structure with $N = 4096$ total subbands, $U = 4000$ usable subbands, and $G = 96$ guard subbands.

[0033] The U usable subbands may be arranged into M interlaces or disjoint subband sets. The M interlaces are disjoint or non-overlapping in that each of the U usable subbands belongs to only one interlace. Each interlace contains S usable subbands, where $U = M \cdot S$. Each interlace may be associated with a different group of $S' = N / M$ subbands that are uniformly distributed across the N total subbands such that consecutive subbands in the group are spaced apart by M subbands. For example, group 1 may contain subbands 1, $M+1$, $2M+1$, and so on, group 2 may contain subbands 2, $M+2$, $2M+2$, and so on, and group M may contain subbands M, 2M, 3M, and so on. For each group, S of the S' subbands are usable subbands and the remaining $S'-S$ subbands are guard subbands. Each interlace may then contain the S usable subbands in the group associated with the interlace. For the exemplary OFDM structure described

above, $M = 8$ interlaces may be formed, with each interlace containing $S = 500$ usable subbands selected from among $S' = 512$ subbands that are evenly spaced apart by $M = 8$ subbands. The S usable subbands in each interlace are thus interlaced with the S usable subbands in each of the other $M - 1$ interlaces.

[0034] In general, the system may utilize any OFDM structure with any number of total, usable, and guard subbands. Any number of interlaces may also be formed. Each interlace may contain any number of usable subbands and any one of the U usable subbands. The interlaces may also contain the same or different numbers of usable subbands. For simplicity, the following description is for the interlaced subband structure shown in FIG. 3 with M interlaces and each interlace containing S uniformly distributed usable subbands. This interlaced subband structure provides several advantages. First, frequency diversity is achieved since each interlace contains usable subbands taken from across the entire system bandwidth. Second, a wireless device may recover data/pilot symbols sent on a given interlace by performing a partial S'-point FFT instead of a full N-point FFT, which can simplify the processing by the wireless device.

[0035] Base station 110 may transmit an FDM pilot on one or more interlaces to allow the wireless devices to perform various functions such as, for example, channel estimation, frequency tracking, time tracking, and so on. Base station 110 may transmit the FDM pilot and traffic data in various manners.

[0036] **FIG. 4A** shows a data and pilot transmission scheme 400 with a "staggered" FDM pilot. In this example, $M = 8$, one interlace is used for the FDM pilot in each symbol period, and the remaining seven interlaces are used for traffic data. The FDM pilot is sent on two designated interlaces in an alternating manner such that pilot symbols are sent on one interlace (e.g., interlace 3) in odd-numbered symbol periods and on another interlace (e.g., interlace 7) in even-numbered symbol periods. The two interlaces used for the FDM pilot are staggered or offset by $M/2 = 4$ interlaces. This staggering allows the wireless devices to observe the channel response for more subbands, which may improve performance.

[0037] **FIG. 4B** shows a data and pilot transmission scheme 410 with a "cycled" FDM pilot. In this example, $M = 8$, one interlace is used for the FDM pilot in each symbol period, and the remaining seven interlaces are used for traffic data. The FDM pilot is sent on all eight interlaces in a cycled manner such that pilot symbols are sent on

a different interlace in each M-symbol period duration. For example, the FDM pilot may be sent on interlace 1 in symbol period 1, then interlace 5 in symbol period 2, then interlace 2 in symbol period 3, and so on, then interlace 8 in symbol period 8, then back to interlace 1 in symbol period 9, and so on. This cycling allows the wireless devices to observe the channel response for all usable subbands.

[0038] In general, an FDM pilot may be sent on any number of interlaces and on any one of the M interlaces in each symbol period. The FDM pilot may also be sent using any pattern, two of which are shown in FIGS. 4A and 4B.

[0039] Base station 110 may transmit the T data streams on the M interlaces in various manners. In a first embodiment, each data stream is sent on the same one or more interlaces in each symbol period in which the data stream is sent. For this embodiment, the interlaces are statically assigned to each data stream. In a second embodiment, each data stream may be sent on different interlaces in different symbol periods in which the data stream is sent. For this embodiment, the interlaces are dynamically assigned to each data stream, which may improve frequency diversity and also ensure that the quality of the channel estimate is independent of the slot index or indices assigned to the data stream. The second embodiment may be viewed as a form of frequency hopping and is described in further detail below.

[0040] To average out channel estimation and detection performance for all T data streams, transmission scheme 410 may be used for the first embodiment with statically assigned interlaces, and either transmission scheme 400 or 410 may be used for the second embodiment with dynamically assigned interlaces. If the FDM pilot is sent on the same one interlace (which is called the pilot interlace) in each symbol period and is used to obtain channel estimates for all M interlaces, then the channel estimate for an interlace that is closer to the pilot interlace is typically better than the channel estimate for an interlace that is further away from the pilot interlace. Detection performance for a data stream may be degraded if the stream is consistently allocated interlaces that are far away from the pilot interlace. The allocation of interlaces with varying distances (or spacing or offset) to the pilot interlace can avoid this performance degradation due to channel estimation bias.

[0041] For the second embodiment, M slots may be defined for each symbol period, and each slot may be mapped to one interlace in one symbol period. A slot usable for traffic data is also called a data slot, and a slot usable for the FDM pilot is also called a pilot slot. The M slots in each symbol period may be given indices 1 through M. Slot

index 1 may be used for the FDM pilot, and slot indices 2 through M may be used for data transmission. The T data streams may be allocated slots with indices 2 through M in each symbol period. The use of slots with fixed indices can simplify the allocation of slots to data streams. The M slot indices may be mapped to the M interlaces in each symbol period based on any mapping scheme that can achieve the desired frequency diversity and channel estimation performance.

[0042] In a first slot-to-interlace mapping scheme, the slot indices are mapped to interlaces in a permutated manner. For transmission scheme 400 with $M = 8$ and one pilot slot and seven data slots in each symbol period, the mapping may be performed as follows. The eight interlaces may be denoted by an original sequence {I_1, I_2, I_3, I_4, I_5, I_6, I_7, I_8}. A permutated sequence may be formed as {I_1, I_5, I_3, I_7, I_2, I_6, I_4, I_8}. The *i*-th interlace in the original sequence is placed in the i_{br}-th position in the permutated sequence, where $i \in \{1 \ldots 8\}$, $i_{br} \in \{1 \ldots 8\}$, and $(i_{br} - 1)$ is a bit-reverse index of $(i-1)$. An offset of -1 is used for i and i_{br} because these indices start from 1 instead of 0. As an example, for $i = 7$, $(i-1) = 6$, the bit representation is '110', the bit-reverse index is '011', $(i_{br} - 1) = 3$, and $i_{br} = 4$. The 7-th interlace in the original sequence is thus placed in the 4-th position in the permutated sequence. The two interlaces used for the FDM pilot are then combined in the permutated sequence to form a shortened interlace sequence {I_1, I_5, $I_{3/7}$, I_2, I_6, I_4, I_8}. The *k*-th slot index used for data transmission (or the *k*-th data slot index), for $k \in \{2 \ldots 8\}$, is then mapped to the $(k-1)$-th interlace in the shortened interlace sequence. For each symbol period thereafter, the shortened interlace sequence is circularly shifted to the right by two positions and wraps around to the left. The *k*-th data slot index is again mapped to the $(k-1)$-th interlace in the circularly shifted shortened interlace sequence.

[0043] **FIG. 5** shows the mapping of slot indices to interlaces for the first mapping scheme described above. Slot index 1, which is used for the FDM pilot, is mapped to interlaces 3 and 7 on alternating symbol periods for transmission scheme 400. Data slot indices 2 through 8 are mapped to the seven interlaces in the shortened interlace sequence {I_1, I_5, $I_{3/7}$, I_2, I_6, I_4, I_8} for the first symbol period, to the circularly or cyclically shifted shortened interlace sequence {I_4, I_8, I_1, I_5, $I_{3/7}$, I_2, I_6} for the second symbol period, and so on. As shown in FIG. 5, each data slot index is mapped to seven different interlaces in seven consecutive symbol periods, where one of the seven

interlaces is either interlace 3 or 7. All seven data slot indices should then achieve similar performance.

[0044] In a second slot-to-interlace mapping scheme, the slot indices are mapped to interlaces in a pseudo-random manner. A pseudo-random number (PN) generator may be used to generate PN numbers that are used to map slot indices to interlaces. The PN generator may be implemented with a linear feedback shift register (LFSR) that implements a particular generator polynomial, e.g., $g(x) = x^{15} + x^{14} + 1$. For each symbol period j, the LFSR is updated and the V least significant bits (LSBs) in the LFSR may be denoted as $\mathrm{PN}(j)$, where $j = 1, 2, \ldots$ and $\mathrm{V} = \log_2 \mathrm{M}$. The k-th data slot index, for $k \in \{2 \ldots \mathrm{M}\}$, may be mapped to interlace $[(\mathrm{PN}(j) + k) \bmod \mathrm{M}] + 1$, if this interlace is not used for the FDM pilot, and to interlace $[(\mathrm{PN}(j) + k + 1) \bmod \mathrm{M}] + 1$, otherwise.

[0045] In a third slot-to-interlace mapping scheme, the slot indices are mapped to interlaces in a circular manner. For each symbol period j, the k-th data slot index, for $k \in \{2 \ldots \mathrm{M}\}$, may be mapped to interlace $[(j + k) \bmod \mathrm{M}] + 1$, if this interlace is not used for the FDM pilot, and to interlace $[(j + k + 1) \bmod \mathrm{M}] + 1$, otherwise.

[0046] The M slot indices may thus be mapped to the M interlaces in various manners. Some exemplary slot-to-interlace mapping schemes have been described above. Other mapping schemes may also be used, and this is within the scope of the invention.

[0047] The slots may be allocated to the T data streams in various manners. In a first slot allocation scheme, each data stream is allocated a sufficient number of slots in each super-frame to transmit a non-negative integer number of data packets (i.e., zero or more data packets). For this scheme, the data packets may be defined to have a fixed size (i.e., a predetermined number of information bits), which can simplify the coding and decoding for the data packets. Each fixed-size data packet may be coded and modulated to generate a coded packet having a variable size that is dependent on the coding and modulation scheme used for the packet. The number of slots needed to transmit the coded packet is then dependent on the coding and modulation scheme used for the packet.

[0048] In a second slot allocation scheme, each data stream may be allocated a non-negative integer number of slots in each super-frame, and an integer number of data packets may be sent in each allocated slot. The same coding and modulation scheme

may be used for all data packets sent in any given slot. Each data packet may have a size that is dependent on (1) the number of data packets being sent in the slot and (2) the coding and modulation scheme used for that slot. For this scheme, the data packets may have variable sizes.

[0049] The slots may also be allocated to the data streams in other manners. For clarity, the following description assumes that the first slot allocation scheme is used by the system.

[0050] Each data stream may be coded in various manners. In an embodiment, each data stream is coded with a concatenated code comprised of an outer code and an inner code. The outer code may be a block code such as a Reed-Solomon (RS) code or some other code. The inner code may be a Turbo code, a convolutional code, or some other code.

[0051] **FIG. 6** shows an exemplary outer coding scheme using a Reed-Solomon outer code. A data stream is partitioned into data packets. In an embodiment, each data packet has a fixed size and contains a predetermined number of information bits or L information bytes (e.g., 1000 bits or 125 bytes). The data packets for the data stream are written into rows of a memory, one packet per row. After K_{rs} data packets have been written into K_{rs} rows, block coding is performed column-wise, one column at a time. In an embodiment, each column contains K_{rs} bytes (one byte per row) and is coded with an (N_{rs}, K_{rs}) Reed-Solomon code to generate a corresponding codeword that contains N_{rs} bytes. The first K_{rs} bytes of the codeword are data bytes (which are also called systematic bytes) and the last $N_{rs} - K_{rs}$ bytes are parity bytes (which may be used by a wireless device for error correction). The Reed-Solomon coding generates $N_{rs} - K_{rs}$ parity bytes for each codeword, which are written to rows $N_{rs} - K_{rs}$ through N_{rs} of the memory after the K_{rs} rows of data. An RS block contains K_{rs} rows of data and $N_{rs} - K_{rs}$ rows of parity. In an embodiment, $N_{rs} = 16$ and K_{rs} is a configurable parameter, e.g., $K_{rs} \in \{12, 14, 16\}$. The Reed-Solomon code is disabled when $K_{rs} = N_{rs}$. Each data/parity packet (or each row) of the RS block is then coded by the Turbo inner code to generate a corresponding coded packet. A code block contains N_{rs} coded packets for the N_{rs} rows of the RS block.

[0052] The N_{rs} coded packets for each code block may be sent in various manners. For example, each code block may be transmitted in one super-frame. Each super-frame may be partitioned into multiple (e.g., four) frames. Each code block may then

be partitioned into multiple (e.g., four) sub-blocks, and each sub-block of the code block may be sent in one frame of the super-frame. The transmission of each code block in multiple parts across a super-frame can provide time diversity.

[0053] Each data stream may be transmitted with or without hierarchical coding, where the term "coding" in this context refers to channel coding rather than data coding at a transmitter. A data stream may be comprised of two substreams, which are called a base stream and an enhancement stream. The base stream may carry base information and may be sent to all wireless devices within the coverage area of the base station. The enhancement stream may carry additional information and may be sent to wireless devices observing better channel conditions. With hierarchical coding, the base stream is coded and modulated to generate a first modulation symbol stream, and the enhancement stream is coded and modulated to generate a second modulation symbol stream. The same or different coding and modulation schemes may be used for the base stream and enhancement stream. The two modulation symbol streams may then be scaled and combined to obtain one data symbol stream.

[0054] Table 1 shows an exemplary set of eight "modes" that may be supported by system 100. These eight modes are given indices of 1 through 8. Each mode is associated with a specific modulation scheme (e.g., QPSK or 16-QAM) and a specific inner code rate (e.g., 1/3, 1/2, or 2/3). The first five modes are for "regular" coding with only the base stream, and the last three modes are for hierarchical coding with the base and enhancement streams. For simplicity, the same modulation scheme and inner code rate are used for both the base and enhancement streams for each hierarchical coding mode.

Table 1

Mode	Modulation Scheme	Inner Code Rate	Number Slots/Packet
1	QPSK	1/3	3
2	QPSK	1/2	2
3	16-QAM	1/3	1.5
4	16-QAM	1/2	1
5	16-QAM	2/3	0.75
6	QPSK/QPSK	1/3	3
7	QPSK/QPSK	1/2	2

8	QPSK/QPSK	2/3	1.5

[0055] The fourth column of Table 1 indicates the number of slots needed to transmit one fixed-size data packet for each mode. Table 1 assumes a data packet size of $2 \cdot S$ information bits and S usable subbands per slot (e.g., $S = 500$). Each slot has a capacity of S data symbols since the slot is mapped to one interlace with S usable subbands and each subband can carry one data symbol. For mode 1, a data packet with $2 \cdot S$ information bits is coded with a rate 1/3 inner code to generate $6 \cdot S$ code bits, which are then mapped to $3 \cdot S$ data symbols using QPSK. The $3 \cdot S$ data symbols for the data packet may be sent in three slots, with each slot carrying S data symbols. Similar processing may be performed for each of the other modes in Table 1.

[0056] Table 1 shows an exemplary design. Data packets of other sizes (e.g., 500 information bits, 2000 information bits, and so on) may also be used. Multiple packet sizes may also be used, for example, so that each packet may be sent in an integer number of slots. For example, a packet size of 1000 information bits may be used for modes 1, 2, and 4, and a packet size of 1333 information bits may be used for modes 3 and 5. In general, the system may also support any number of modes for any number of coding and modulation schemes, any number of data packet sizes, and any packet size.

[0057] **FIG. 7A** shows transmission of a minimum integer number of data packets, using one slot in each of an integer number of symbol periods, for each of the first five modes listed in Table 1. One data packet may be sent using one slot in (1) three symbol periods for mode 1, (2) two symbol periods for mode 2, and (3) one symbol period for mode 4. Two data packets may be sent using one slot in three symbol periods for mode 3, since each data packet takes 1.5 slots to send. Four data packets may be sent using one slot in three symbol periods for mode 5, since each data packet takes 0.75 slots to send.

[0058] **FIG. 7B** shows transmission of a minimum integer number of data packets, using an integer number of slots in one symbol period, for each of the first five modes listed in Table 1. One data packet may be sent in one symbol period using (1) three slots for mode 1, (2) two slots for mode 2, and (3) one slot for mode 4. Two data packets may be sent in one symbol period using three slots for mode 3. Four data packets may be sent in one symbol period using three slots for mode 5.

[0059] As shown in FIGS. 7A and 7B, the minimum number of data packets may be transmitted in several manners for each mode (except for mode 4). Transmitting the minimum number of data packets in a shorter time period reduces the amount of ON time required to receive the data packets but provides less time diversity. The converse is true for transmitting the minimum number of data packets over a longer time period.

[0060] **FIG. 8A** shows the partitioning of a single coded packet into three slots for mode 1. The three slots may be for three different interlaces in one symbol period or one interlace in three different symbol periods. The three slots may observe different channel conditions. The bits in the coded packet may be interleaved (i.e., reordered) prior to the partitioning into three slots. The interleaving for each coded packet can randomize the signal-to-noise ratios (SNRs) of the bits across the coded packet, which may improve decoding performance. The interleaving may be performed in various manners, as is known in the art. The interleaving may also be such that adjacent bits in the coded packet are not sent in the same data symbol.

[0061] **FIG. 8B** shows the partitioning of four coded packets into three slots for mode 5. The three slots may be sequentially filled by the four coded packets, as shown in FIG. 8B. When multiple coded packets share a slot (such as for modes 3 and 5), all of the bits to be sent in the slot may be interleaved such that the bits for each coded packet sent in the slot are distributed across the subbands used for the slot. The interleaving across each slot provides frequency diversity for each coded packet sent in the slot and may improve decoding performance.

[0062] The interleaving across a slot may be performed in various manners. In an embodiment, the bits for all coded packets to be sent in a given slot are first mapped to data symbols, and the data symbols are then mapped to the subbands used for the slot in a permutated manner. For the symbol-to-subband mapping, a first sequence with S' sequential values, 0 through $S'-1$, is initially formed. A second sequence of S' values is then created such that the i-th value in the second sequence is equal to the bit reverse of the i-th value in the first sequence. All values that are equal to or greater than S' in the second sequence are removed to obtain a third sequence with S values ranging from 0 through $S-1$. Each value in the third sequence is then incremented by one to obtain a sequence of S permutated index values ranging from 1 through S, which is denoted as $F(j)$. The j-th data symbol in the slot may be mapped to the $F(j)$-th subband in the interlace used for the slot. For example, if $S=500$ and $S'=512$, then the first sequence

is {0, 1, 2, 3, ..., 510, 511}, the second sequence is {0, 256, 128, 384, ..., 255, 511}, and the third sequence is {0, 256, 128, 384, ... 255}. The sequence $F(j)$ only needs to be computed once and may be used for all slots. Other mapping schemes may also be used for the symbol-to-subband mapping to achieve interleaving across each slot.

[0063] In general, each data stream may carry any number of data packets in each super-frame, depending on the data rate of the stream. Each data stream is allocated a sufficient number of slots in each super-frame based on its data rate, subject to the availability of slots and possibly other factors. For example, each data stream may be constrained to a specified maximum number of slots in each symbol period, which may be dependent on the mode used for the data stream. Each data stream may be limited to a specified maximum data rate, which is the maximum number of information bits that may be transmitted in each symbol period for the data stream. The maximum data rate is typically set by the decoding and buffering capabilities of the wireless devices. Constraining each data stream to be within the maximum data rate ensures that the data stream can be recovered by wireless devices having the prescribed decoding and buffering capabilities. The maximum data rate limits the number of data packets that may be transmitted in each symbol period for the data stream. The maximum number of slots may then be determined by the maximum number of data packets and the mode used for the data stream.

[0064] In an embodiment, each data stream may be allocated an integer number of slots in any given symbol period, and multiple data streams do not share an interlace. For this embodiment, up to $M-1$ data streams may be sent on the $M-1$ data slots in each symbol period, assuming that one slot is used for the FDM pilot. In another embodiment, multiple data streams may share an interlace.

[0065] **FIG. 9A** shows a block diagram of an embodiment of TX data processor 120 at base station 110. TX data processor 120 includes T TX data stream processors 910a through 910t for the T data streams, a TX overhead data processor 930 for overhead/control data, a pilot processor 932 for the TDM and FDM pilots, and a multiplexer (Mux) 940. Each TX data stream processor 910 processes a respective data stream $\{d_i\}$ to generate a corresponding data symbol stream $\{Y_i\}$, for $i \in \{1 \ldots T\}$.

[0066] Within each TX data stream processor 910, an encoder 912 receives and encodes data packets for its data stream $\{d_i\}$ and provides coded packets. Encoder 912 performs encoding in accordance with, for example, a concatenated code comprised of a

Reed-Solomon outer code and a Turbo or convolutional inner code. In this case, encoder 912 encodes each block of K_{rs} data packets to generate N_{rs} coded packets, as shown in FIG. 6. The encoding increases the reliability of the transmission for the data stream. Encoder 912 may also generate and append a cyclic redundancy check (CRC) value to each coded packet, which may be used by a wireless device for error detection (i.e., to determine whether the packet is decoded correctly or in error). Encoder 912 may also shuffle the coded packets.

[0067] An interleaver 914 receives the coded packets from encoder 912 and interleaves the bits in each coded packet to generate an interleaved packet. The interleaving provides time and/or frequency diversity for the packet. A slot buffer 916 is then filled with interleaved packets for all the slots allocated to the data stream, e.g., as shown in FIG. 8A or 8B.

[0068] A scrambler 918 receives and scrambles the bits for each slot with a PN sequence to randomize the bits. M different PN sequences may be used for the M slot indices. The M PN sequences may be generated, for example, with a linear feedback shift register (LFSR) that implements a particular generator polynomial, e.g., $g(x) = x^{15} + x^{14} + 1$. The LFSR may be loaded with a different 15-bit initial value for each slot index. Furthermore, the LFSR may be reloaded at the start of each symbol period. Scrambler 918 may perform an exclusive-OR on each bit in a slot with a bit in the PN sequence to generate a scrambled bit.

[0069] A bit-to-symbol mapping unit 920 receives the scrambled bits for each slot from scrambler 918, maps these bits to modulation symbols in accordance with a modulation scheme (e.g., QPSK or 16-QAM) selected for the data stream, and provides data symbols for the slot. The symbol mapping may be achieved by (1) grouping sets of B bits to form B-bit binary values, where $B \geq 1$, and (2) mapping each B-bit binary value to a complex value for a point in a signal constellation for the modulation scheme. The outer and inner codes for encoder 912 and the modulation scheme for mapping unit 920 are determined by the mode used for the data stream.

[0070] If the data stream is sent using hierarchical coding, then the base stream may be processed by one set of processing units 912 through 920 to generate a first stream of modulation symbols, and the enhancement stream may be processed by another set of processing units 912 through 920 to generate a second stream of modulation symbols (not shown in FIG. 9 for simplicity). The same coding and modulation scheme may be

used for both the base stream and the enhancement stream, as shown in Table 1, or different coding and modulation schemes may be used for the two streams. A combiner may then receive and combine the first and second modulation symbol streams to generate the data symbols for the data stream. The hierarchical coding may also be performed in other manners. For example, the scrambled bits for both the base stream and enhancement stream may be provided to a single bit-to-symbol mapping unit that provides the data symbols for the data stream.

[0071] A slot-to-interlace mapping unit 922 maps each slot assigned to data stream $\{d_i\}$ to the proper interlace based on the slot-to-interlace mapping scheme used by the system (e.g., as shown in FIG. 5). A symbol-to-subband mapping unit 924 then maps the S data symbols in each slot to the proper subbands in the interlace to which the slot is mapped. The symbol-to-subband mapping may be performed in a manner to distribute the S data symbols across the S subbands used for the slot, as described above. Mapping unit 924 provides data symbols for data stream $\{d_i\}$, which are mapped to the proper subbands used for the data stream.

[0072] TX overhead data processor 930 processes overhead/control data in accordance with a coding and modulation scheme used for overhead/control data and provides overhead symbols. Pilot processor 932 performs processing for the TDM and FDM pilots and provides pilot symbols. Multiplexer 940 receives the mapped data symbols for the T data streams from TX data stream processors 910a through 910t, the overhead symbols from TX overhead data processor 930, the pilot symbols from pilot processor 932, and guard symbols. Multiplexer 940 provides the data symbols, overhead symbols, pilot symbols, and guard symbols onto the proper subbands and symbol periods based on a MUX_TX control from controller 140 and outputs a composite symbol stream, $\{Y_C\}$.

[0073] **FIG. 9B** shows a block diagram of an embodiment of modulator 130 at base station 110. Modulator 130 includes an inverse fast Fourier transform (IFFT) unit 950 and a cyclic prefix generator 952. For each symbol period, IFFT unit 950 transforms the N symbols for the N total subbands to the time domain with an N-point IFFT to obtain a "transformed" symbol that contains N time-domain samples. To combat intersymbol interference (ISI), which is caused by frequency selective fading, cyclic prefix generator 952 repeats a portion (or C samples) of each transformed symbol to form a corresponding OFDM symbol that contains $N+C$ samples. The repeated

portion is often called a cyclic prefix or guard interval. For example, the cyclic prefix length may be $C = 512$ for $N = 4096$. Each OFDM symbol is transmitted in one OFDM symbol period (or simply, symbol period), which is $N + C$ sample periods. Cyclic prefix generator 952 provides an output sample stream $\{y\}$ for the composite symbol stream $\{Y_C\}$.

[0074] **FIG. 10A** shows a block diagram of an embodiment of demodulator 160 at wireless device 150. Demodulator 160 includes a cyclic prefix removal unit 1012, a Fourier transform unit 1014, a channel estimator 1016, and a detector 1018. Cyclic prefix removal unit 1012 removes the cyclic prefix in each received OFDM symbol and provides a sequence of N input samples, $\{x(n)\}$, for the received OFDM symbol. Fourier transform unit 1014 performs a partial Fourier transform on the input sample sequence $\{x(n)\}$ for each selected interlace m and provides a set of S received symbols, $\{X_m(k)\}$, for that interlace, where $m = 1 \ldots M$. Channel estimator 1016 derives channel gain estimate $\{\hat{H}_m(k)\}$ for each selected interlace m based on the input sample sequence $\{x(n)\}$. Detector 1018 performs detection (e.g., equalization or matched filtering) on the set of S received symbols $\{X_m(k)\}$ for each selected interlace with the channel gain estimate $\{\hat{H}_m(k)\}$ for that interlace and provides S detected data symbols $\{\hat{Y}_m(k)\}$ for the interlace.

[0075] **FIG. 10B** shows a block diagram of an embodiment of RX data processor 170 at wireless device 150. A multiplexer 1030 receives the detected data symbols for all interlaces from detector 1018, performs multiplexing of the detected data and overhead symbols for each symbol period based on the MUX_RX control, provides each detected data symbol stream of interest to a respective RX data stream processor 1040, and provides a detected overhead symbol stream to an RX overhead data processor 1060.

[0076] Within each RX data stream processor 1040, a subband-to-symbol demapping unit 1042 maps the received symbol on each subband in a selected interlace to the proper position within a slot. An interlace-to-slot demapping unit 1044 maps each selected interlace to the proper slot. A symbol-to-bit demapping unit 1046 maps the received symbols for each slot to code bits. A descrambler 1048 descrambles the code bits for each slot and provides descrambled data. A slot buffer 1050 buffers one or more slots of descrambled data, performs reassembly of packets as needed, and provides

descrambled packets. A deinterleaver 1052 deinterleaves each descrambled packet and provides a deinterleaved packet. A decoder 1054 decodes the deinterleaved packets and provides decoded data packets for data stream $\{d_i\}$. In general, the processing performed by the units within RX data stream processor 1040 is complementary to the processing performed by the corresponding units within TX data stream processor 910 in FIG. 9A. The symbol-to-bit demapping and the decoding are performed in accordance with the mode used for the data stream. RX overhead data processor 1060 processes the received overhead symbols and provides decoded overhead data.

[0077] Because of the periodic structure of the M interlaces, Fourier transform unit 1014 may perform a partial S′-point Fourier transform for each selected interlace m to obtain the set of S received symbols $\{X_m(k)\}$ for that interlace. The Fourier transform for the S′ subbands that include all S subbands of interlace m, where $m = 1 \ldots \mathrm{M}$, may be expressed as:

$$\begin{aligned} X(\mathrm{M}\cdot k+m) &= \sum_{n=1}^{\mathrm{N}} x(n)\cdot W_{\mathrm{N}}^{(\mathrm{M}\cdot k+m)n} , \\ &= \sum_{n=1}^{\mathrm{N}} x(n)\cdot W_{\mathrm{N}}^{mn}\cdot W_{\mathrm{S}'}^{kn} , \end{aligned} \qquad \text{for } k = 1 \ldots \mathrm{S}', \qquad \text{Eq (1)}$$

where $x(n)$ is the input sample for sample period n, $W_{\mathrm{N}}^{\alpha\beta} = e^{-j\frac{2\pi(\alpha-1)(\beta-1)}{\mathrm{N}}}$, and $\mathrm{N} = \mathrm{M}\cdot\mathrm{S}'$. The following terms may be defined:

$$\tilde{x}_m(n) = x(n)\cdot W_{\mathrm{N}}^{mn} , \qquad \text{for } n = 1 \ldots \mathrm{N}, \text{ and} \qquad \text{Eq (2)}$$

$$g_m(n) = \sum_{i=0}^{\mathrm{M}-1} \tilde{x}_m(\mathrm{S}'\cdot i+n) , \qquad \text{for } n = 1 \ldots \mathrm{S}' , \qquad \text{Eq (3)}$$

where $\tilde{x}_m(n)$ is a rotated sample obtained by rotating the input sample $x(n)$ by $W_{\mathrm{N}}^{mn} = e^{-j\frac{2\pi(m-1)(n-1)}{\mathrm{N}}}$, which is a phasor that varies from sample to sample (the -1 in the exponent for the terms $m-1$ and $n-1$ is due to an index numbering scheme that starts with 1 instead of 0); and

$g_m(n)$ is a time-domain value obtained by accumulating M rotated samples that are spaced apart by S′ samples.

[0078] Equation (1) may then be expressed as:

$$X_m(k) = X(\mathrm{M} \cdot k + m) = \sum_{n=1}^{\mathrm{S}'} g_m(n) \cdot W_{\mathrm{S}'}^{kn} \quad , \qquad \text{for } k = 1 \ \ldots \ \mathrm{S}' \ . \qquad \text{Eq (4)}$$

[0079] A partial S′-point Fourier transform for interlace m may be performed as follows. Each of the N input samples in the sequence $\{x(n)\}$ for one symbol period is first rotated by W_{N}^{mn}, as shown in equation (2), to obtain a sequence of N rotated samples $\{\widetilde{x}_m(n)\}$. The rotated samples are then accumulated, in S′ sets of M rotated samples, to obtain S′ time-domain values $\{g_m(n)\}$, as shown in equation (3). Each set contains every S′-th rotated sample in the sequence $\{\widetilde{x}_m(n)\}$, with the S′ sets being associated with different starting rotated samples in the sequence $\{\widetilde{x}_m(n)\}$. A normal S′-point Fourier transform is then performed on the S′ time-domain values $\{g_m(n)\}$ to obtain the S′ received symbols for interlace m. The received symbols for the S usable subbands are retained, and the received symbols for the S′-S unused subbands are discarded.

[0080] For channel estimation, a partial S′-point Fourier transform may be performed on the N input samples for interlace p used for the FDM pilot to obtain a set of S received pilot symbols, $\{X_p(k)\}$ or $X(\mathrm{M} \cdot k + p)$. The modulation on the received pilot symbols is then removed to obtain channel gain estimates $\{\hat{H}_p(k)\}$ for the subbands in interlace p, as follows:

$$\hat{H}_p(k) = \hat{H}(\mathrm{M} \cdot k + p) = X(\mathrm{M} \cdot k + p) \cdot P^*(\mathrm{M} \cdot k + p) \ , \quad \text{for } k = 1 \ \ldots \ \mathrm{S}', \qquad \text{Eq (5)}$$

where $P(\mathrm{M} \cdot k + p)$ is the known pilot symbol for the k-th subband in interlace p and " * " is a complex conjugate. Equation (5) assumes that all S′ subbands are used for pilot transmission. An S′-point IFFT is then performed on the channel gain estimates $\{\hat{H}_p(k)\}$ to obtain a sequence of S′ modulated time-domain channel gain values, $\{h_p(n)\}$, which may be expressed as: $h_p(n) = h(n) \cdot W_{\mathrm{N}}^{pn}$, for $n = 1 \ \ldots \ \mathrm{S}'$. The channel gain values in the sequence $\{h_p(n)\}$ are then derotated by multiplication with W_{N}^{-pn} to obtain a sequence of S′ derotated time-domain channel gain values, $h(n) = h_p(n) \cdot W_{\mathrm{N}}^{-pn}$, for $n = 1 \ \ldots \ \mathrm{S}'$.

[0081] The channel gain estimates for the subbands in interlace m may then be expressed as:

$$\begin{aligned}\hat{H}_m(k) &= \hat{H}(\mathrm{M}\cdot k+m) \quad , \\ &= \sum_{n=1}^{\mathrm{S}'} h(n)\cdot W_{\mathrm{N}}^{(\mathrm{M}\cdot k+m)n} \quad , \qquad \text{for } k=1 \ \ldots \ \mathrm{S}'. \\ &= \sum_{n=1}^{\mathrm{S}'} h(n)\cdot W_{\mathrm{N}}^{mn}\cdot W_{\mathrm{S}'}^{kn} \quad ,\end{aligned} \qquad \text{Eq (6)}$$

As indicated in equation (6), the channel gain estimates for the subbands in interlace m may be obtained by first multiplying each derotated time-domain channel gain value in the sequence $\{h(n)\}$ by W_{N}^{mn} to obtain a sequence of S′ rotated channel gain values, $\{\widetilde{h}_m(n)\}$. A normal S′-point FFT is then performed on the sequence $\{\widetilde{h}_m(n)\}$ to obtain S′ channel gain estimates for the subbands in interlace m. The derotation of $h_p(n)$ by W_{N}^{-pn} and the rotation of $h(n)$ by W_{N}^{mn} may be combined, so that the rotated channel gain values for interlace m may be obtained as $h_m(n)=h_p(n)\cdot W_{\mathrm{N}}^{(m-p)n}$, for $n=1 \ \ldots \ \mathrm{S}'$.

[0082] An exemplary channel estimation scheme has been described above. The channel estimation may also be performed in other manners. For example, the channel estimates obtained for different interlaces used for pilot transmission may be filtered (e.g., over time) and/or post-processed (e.g., based on a least square estimate of the impulse response $\{h(n)\}$) to obtain a more accurate channel estimate for each interlace of interest.

[0083] The multiplexing techniques described herein may be implemented by various means. For example, these techniques may be implemented in hardware, software, or a combination thereof. For a hardware implementation, the processing units used to perform the multiplexing at a base station may be implemented within one or more application specific integrated circuits (ASICs), digital signal processors (DSPs), digital signal processing devices (DSPDs), programmable logic devices (PLDs), field programmable gate arrays (FPGAs), processors, controllers, micro-controllers, microprocessors, other electronic units designed to perform the functions described herein, or a combination thereof. The processing units used to perform the complementary processing at a wireless device may also be implemented within one or more ASICs, DSPs, and so on.

[0084] For a software implementation, the multiplexing techniques may be implemented with modules (e.g., procedures, functions, and so on) that perform the functions described herein. The software codes may be stored in a memory unit (e.g., memory unit 142 or 182 in FIG. 1) and executed by a processor (e.g., controller 140 or 180). The memory unit may be implemented within the processor or external to the processor, in which case it can be communicatively coupled to the processor via various means as is known in the art.

[0085] The previous description of the disclosed embodiments is provided to enable any person skilled in the art to make or use the present invention. Various modifications to these embodiments will be readily apparent to those skilled in the art, and the generic principles defined herein may be applied to other embodiments without departing from the spirit or scope of the invention. Thus, the present invention is not intended to be limited to the embodiments shown herein but is to be accorded the widest scope consistent with the principles and novel features disclosed herein.

[0086] **WHAT IS CLAIMED IS:**

CLAIMS

1. A method of transmitting data in a wireless multi-carrier communication system, comprising:

allocating slots to each of a plurality of data symbol streams, wherein each slot is a unit of transmission and a plurality of slots are frequency division multiplexed in each symbol period;

multiplexing data symbols in each data symbol stream onto the slots allocated to the data symbol stream; and

forming a composite symbol stream with multiplexed data symbols for the plurality of data symbol streams, wherein the plurality of data symbol streams are independently recoverable by a receiver.

2. The method of claim 1, further comprising:

forming a plurality of non-overlapping interlaces with U frequency subbands usable for transmission, where $U > 1$ and each interlace is a different set of frequency subbands selected from among the U frequency subbands; and

mapping the plurality of slots in each symbol period to the plurality of interlaces.

3. The method of claim 1, further comprising:

forming 2^N non-overlapping interlaces with a plurality of frequency subbands usable for transmission, where $N \geq 1$ each interlace is a different set of frequency subbands selected from among the plurality of frequency subbands; and

mapping the plurality of slots in each symbol period to the 2^N interlaces.

4. The method of claim 3, wherein N is equal to 1, 2, 3 or 4.

5. The method of claim 2, wherein the forming the plurality of non-overlapping interlaces comprises

forming the plurality of interlaces with equal number of frequency subbands.

6. The method of claim 2, wherein the forming the plurality of non-overlapping interlaces comprises

forming the plurality of interlaces with the frequency subbands in each interlace being interlaced with the frequency subbands in each of remaining interlaces.

7. The method of claim 2, wherein the forming the plurality of non-overlapping interlaces comprises

forming a plurality of groups of frequency subbands, each group including frequency subbands uniformly distributed across T total frequency subbands in the system, where T ≥ U, and

forming each interlace with frequency subbands selected from a respective group of frequency subbands.

8. The method of claim 2, wherein the allocating slots to each of the plurality of data symbol streams comprises

allocating each of the plurality of interlaces to one data symbol stream, if at all, in each symbol period.

9. The method of claim 2, wherein the plurality of slots in each symbol period are identified by slot indices, the method further comprising:

for each symbol period, mapping the slot indices to the plurality of interlaces based on a mapping scheme.

10. The method of claim 9, wherein the mapping the slot indices to the plurality of interlaces comprises

mapping each slot index used for data transmission to different ones of the plurality of interlaces in different symbol periods.

11. The method of claim 2, further comprising:

distributing data symbols multiplexed onto each allocated slot across the frequency subbands in the interlace to which the slot is mapped.

12. The method of claim 11, wherein the distributing the data symbols multiplexed onto each allocated slot comprises

distributing data symbols for each data packet sent in the slot across the frequency subbands in the interlace to which the slot is mapped.

13. The method of claim 2, further comprising:

selecting slots for pilot transmission from among the plurality of slots in each symbol period; and

multiplexing pilot symbols onto the slots used for pilot transmission.

14. The method of claim 13, further comprising:

mapping the slots used for pilot transmission to different interlaces in different symbol periods.

15. The method of claim 13, further comprising:

mapping the plurality of slots in each symbol period to the plurality of interlaces such that interlaces used for pilot transmission have varying distances to interlaces used for data transmission.

16. The method of claim 9, further comprising:

allocating at least one slot index for pilot transmission; and

allocating remaining slot indices for data transmission.

17. The method of claim 16, further comprising:

mapping the at least one slot index used for pilot transmission to at least one predetermined interlace; and

mapping each slot index used for data transmission to different interlaces in different symbol periods.

18. The method of claim 1, further comprising:

processing a plurality of data streams to obtain the plurality of data symbol streams, one data symbol stream for each data stream.

19. The method of claim 1, wherein the allocating the slots to each of the plurality of data symbol streams comprises

allocating a particular number of slots to each data symbol stream based on at least one packet size and at least one coding and modulation scheme used for the data symbol stream.

20. The method of claim 18, wherein the processing the plurality of data streams comprises

encoding data packets for each data stream in accordance with a coding scheme to generate coded packets for the data stream; and

modulating the coded packets for each data stream in accordance with a modulation scheme to generate data symbols for the corresponding data symbol stream.

21. The method of claim 18,

wherein the encoding the data packets for each data stream comprises encoding an integer number of data packets for each data stream in each frame of a predetermined time period, and

wherein the allocating the slots to each of the plurality of data symbol streams comprises allocating an integer number of slots to each data symbol stream in each frame based on the number of data packets being transmitted in the frame for the corresponding data stream.

22. The method of claim 1, wherein the allocating the slots to each of the plurality of data symbol streams comprises

allocating each data symbol stream a particular number of slots determined by decoding constraint and a coding and modulation scheme used for the data symbol stream.

23. An apparatus in a wireless multi-carrier communication system, comprising:

a controller operative to allocate slots to each of a plurality of data symbol streams, wherein each slot is a unit of transmission and a plurality of slots are frequency division multiplexed in each symbol period; and

a data processor operative to multiplex data symbols in each data symbol stream onto the slots allocated to the data symbol stream and to form a composite symbol

stream with multiplexed data symbols for the plurality of data symbol streams, wherein the plurality of data symbol streams are independently recoverable by a receiver.

24. The apparatus of claim 23, wherein the controller is further operative to form a plurality of non-overlapping interlaces with U frequency subbands usable for transmission, where $U > 1$, and to map the plurality of slots in each symbol period to the plurality of interlaces, each interlace being a different set of frequency subbands selected from among the U frequency subbands.

25. The apparatus of claim 24, wherein the plurality of slots in each symbol period are identified by slot indices, and wherein the data processor is further operative to, for each symbol period, map the slot indices to the plurality of interlaces based on a mapping scheme.

26. The apparatus of claim 23, wherein the controller is further operative to select slots for pilot transmission from among the plurality of slots in each symbol period, and wherein the data processor is further operative to multiplex pilot symbols onto the slots used for pilot transmission.

27. The apparatus of claim 23, wherein the controller is further operative to allocate a particular number slots to each data symbol stream based on at least one packet size and at least one coding and modulation scheme used for the data symbol stream.

28. The apparatus of claim 23, the data processor is further operative to process a plurality of data streams to obtain the plurality of data symbol streams, one data symbol stream for each data stream.

29. The apparatus of claim 23, wherein the wireless multi-carrier communication system utilizes orthogonal frequency division multiplexing (OFDM).

30. The apparatus of claim 23, wherein the wireless multi-carrier communication system is a broadcast system.

31. An apparatus in a wireless multi-carrier communication system, comprising:

means for allocating slots to each of a plurality of data symbol streams, wherein each slot is a unit of transmission and a plurality of slots are frequency division multiplexed in each symbol period;

means for multiplexing data symbols in each data symbol stream onto the slots allocated to the data symbol stream; and

means for forming a composite symbol stream with multiplexed data symbols for the plurality of data symbol streams, wherein the plurality of data symbol streams are independently recoverable by a receiver.

32. The apparatus of claim 31, further comprising:

means for forming a plurality of non-overlapping interlaces with U frequency subbands usable for transmission, where $U > 1$ and each interlace is a different set of frequency subbands selected from among the U frequency subbands; and

means for mapping the plurality of slots in each symbol period to the plurality of interlaces.

33. The apparatus of claim 32, wherein the plurality of slots in each symbol period are identified by slot indices, the apparatus further comprising:

means for mapping the slot indices to the plurality of interlaces for each symbol period based on a mapping scheme.

34. The apparatus of claim 31, further comprising:

means for selecting slots for pilot transmission from among the plurality of slots in each symbol period; and

means for multiplexing pilot symbols onto the slots used for pilot transmission.

35. The apparatus of claim 31, further comprising:

means for processing a plurality of data streams to obtain the plurality of data symbol streams, one data symbol stream for each data stream.

36. A method of receiving data in a wireless multi-carrier communication system, comprising:

selecting at least one data stream for recovery from among a plurality of data streams transmitted by a transmitter in the system;

determining slots used for each selected data stream, wherein each slot is a unit of transmission and a plurality of slots are frequency division multiplexed in each symbol period, wherein data symbols for each of the plurality of data streams are multiplexed onto slots allocated to the data stream, and wherein the plurality of data streams are independently recoverable by a receiver;

multiplexing detected data symbols obtained for slots used for each selected data stream onto a detected data symbol stream, wherein each detected data symbol is an estimate of a data symbol and at least one detected data symbol stream is obtained for the at least one data stream selected for recovery; and

processing each detected data symbol stream to obtain a corresponding decoded data stream.

37. The method of claim 35, further comprising:

mapping the plurality of slots in each symbol period to a plurality of non-overlapping interlaces formed with U frequency subbands usable for transmission, where U > 1 and each interlace is a different set of frequency subbands selected from among the U frequency subbands,.

38. The method of claim 37, wherein the plurality of slots in each symbol period are identified by slot indices, and wherein the mapping the plurality of slots in each symbol period comprises

mapping the slot indices to the plurality of interlaces in each symbol period based on a mapping scheme.

39. The method of claim 36, further comprising:

performing a partial Fourier transform for each slot used for each selected data stream to obtain received data symbols for the slot, the partial Fourier transform being a Fourier transform for fewer than all frequency subbands in the system; and

performing detection on the received data symbols for each slot used for each selected data stream to obtain detected symbols for the slot.

40. The method of claim 36, further comprising:

performing a partial Fourier transform for each slot used for pilot transmission to obtain a channel estimate for the slot.

41. The method of claim 40, further comprising:

deriving a channel estimate for each slot used for each selected data stream based on channel estimates obtained from slots used for pilot transmission.

42. An apparatus in a wireless multi-carrier communication system, comprising:

a controller operative to select at least one data stream for recovery from among a plurality of data streams transmitted by a transmitter in the system and to determine slots used for each selected data stream, wherein each slot is a unit of transmission and a plurality of slots are frequency division multiplexed in each symbol period, wherein data symbols for each of the plurality of data streams are multiplexed onto slots allocated to the data stream, and wherein the plurality of data streams are independently recoverable by a receiver; and

a data processor operative to multiplex detected data symbols obtained for slots used for each selected data stream onto a detected data symbol stream and to process each detected data symbol stream to obtain a corresponding decoded data stream, wherein each detected data symbol is an estimate of a data symbol and at least one detected data symbol stream is obtained for the at least one data stream selected for recovery.

43. The apparatus of claim 42, wherein the controller is further operable to map the plurality of slots in each symbol period to a plurality of non-overlapping interlaces formed with U frequency subbands usable for transmission, where $U > 1$ and each interlace is a different set of frequency subbands selected from among the U frequency subbands.

44. The apparatus of claim 42, further comprising:

a demodulator operative to perform a partial Fourier transform for each slot used for each selected data stream to obtain received data symbols for the slot and to perform detection on the received data symbols for each slot used for each selected data stream to obtain detected symbols for the slot.

45. An apparatus in a wireless multi-carrier communication system, comprising:

means for selecting at least one data stream for recovery from among a plurality of data streams transmitted by a transmitter in the system;

means for determining slots used for each selected data stream, wherein each slot is a unit of transmission and a plurality of slots are frequency division multiplexed in each symbol period, wherein data symbols for each of the plurality of data streams are multiplexed onto slots allocated to the data stream, and wherein the plurality of data streams are independently recoverable by a receiver;

means for multiplexing detected data symbols obtained for slots used for each selected data stream onto a detected data symbol stream, wherein each detected data symbol is an estimate of a data symbol and at least one detected data symbol stream is obtained for the at least one data stream selected for recovery; and

means for processing each detected data symbol stream to obtain a corresponding decoded data stream.

46. The apparatus of claim 45, further comprising:

means for mapping the plurality of slots in each symbol period to a plurality of non-overlapping interlaces formed with U frequency subbands usable for transmission, where $U > 1$ and each interlace is a different set of frequency subbands selected from among the U frequency subbands.

47. The apparatus of claim 45, further comprising:

means for performing a partial Fourier transform for each slot used for each selected data stream to obtain received data symbols for the slot; and

means for performing detection on the received data symbols for each slot used for each selected data stream to obtain detected symbols for the slot.

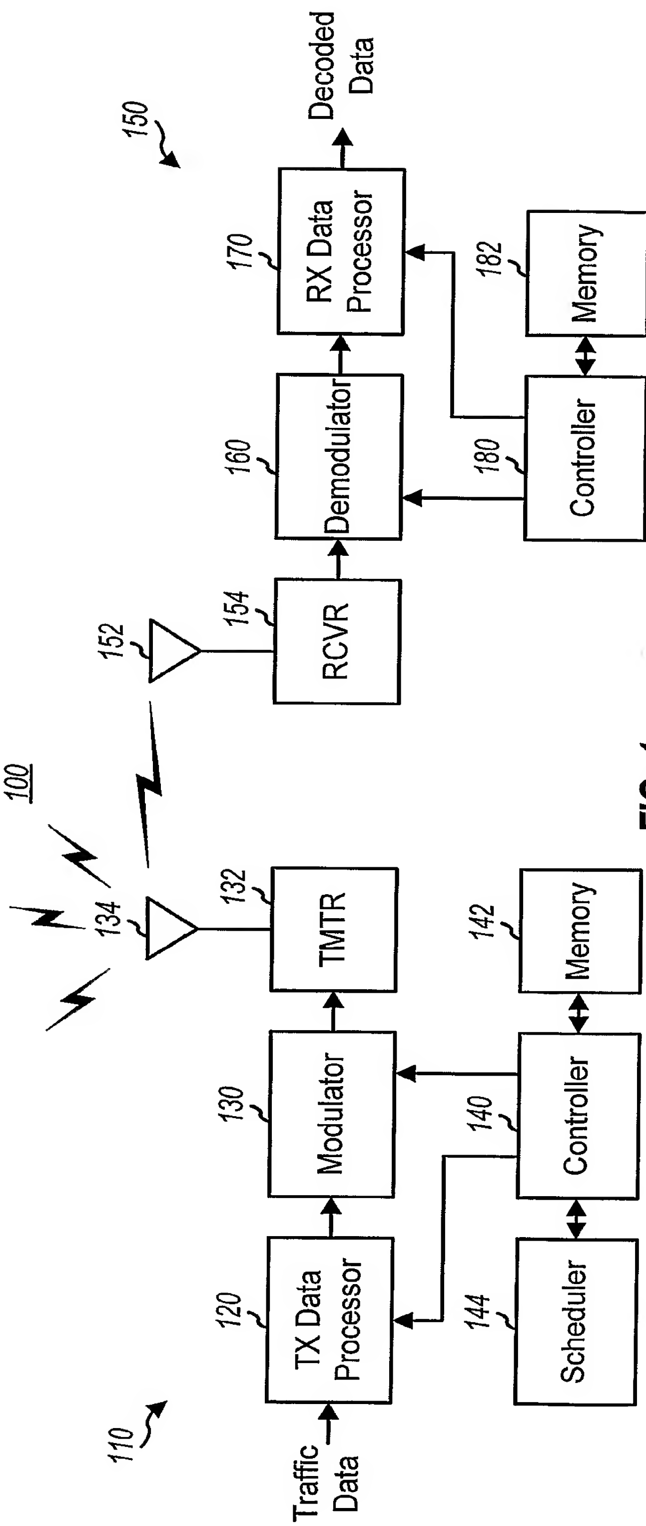

100
110
Traffic Data
TX Data Processor
120
Modulator
130
TMTR
132
134
Controller
140
Memory
142
Scheduler
144
150
152
RCVR
154
Demodulator
160
RX Data Processor
170
Decoded Data
Controller
180
Memory
182

FIG. 1

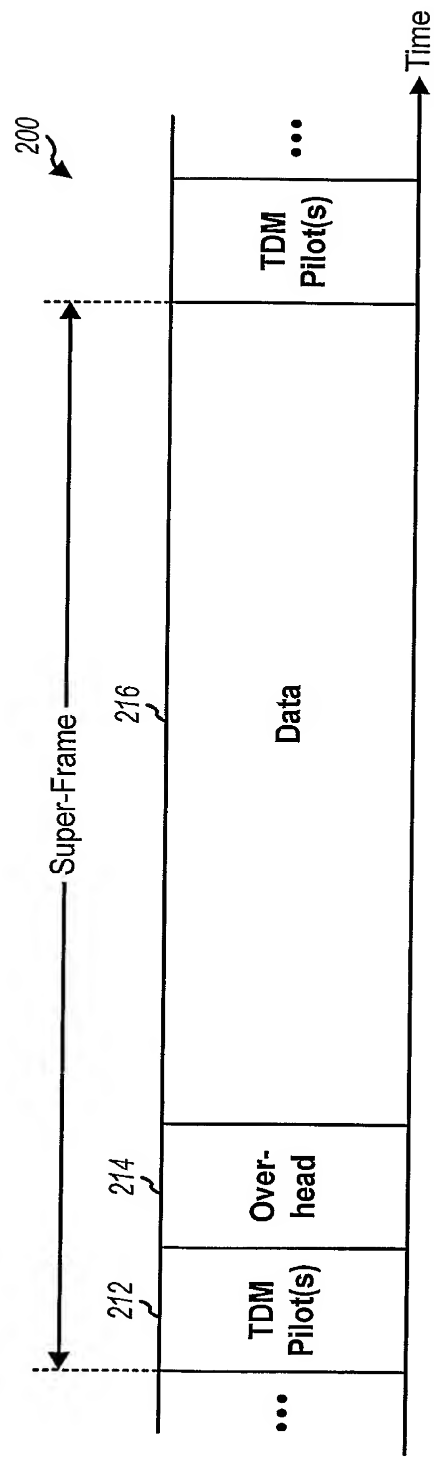

200
Super-Frame
TDM Pilot(s)
212
Over-head
214
Data
216
TDM Pilot(s)
Time

FIG. 2

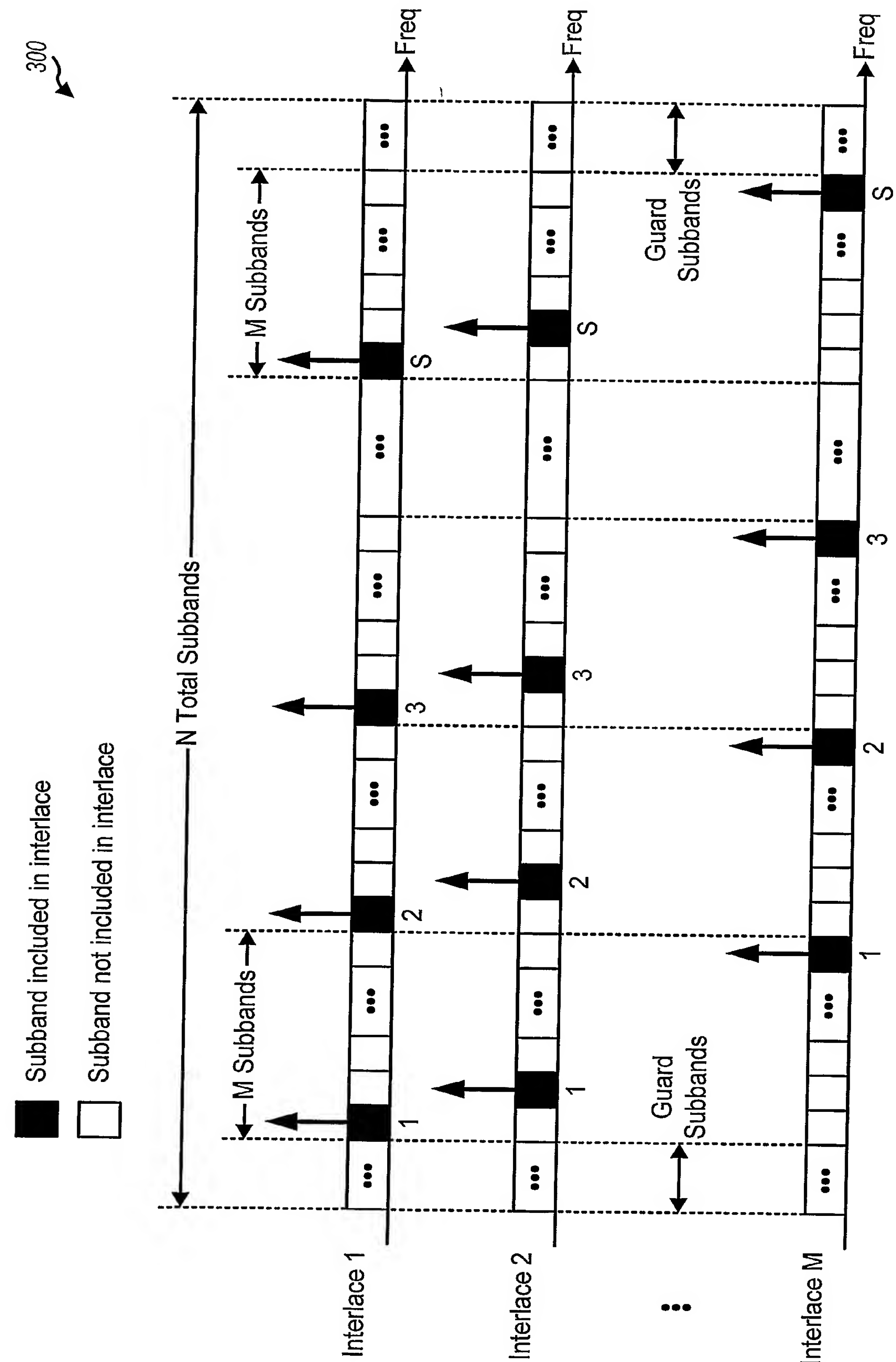

300
Subband included in interlace
Subband not included in interlace
N Total Subbands
M Subbands
M Subbands
Guard Subbands
Guard Subbands
Interlace 1
Interlace 2
Interlace M
Freq
Freq
Freq
1
2
3
S
1
2
3
S
1
2
3
S

FIG. 3

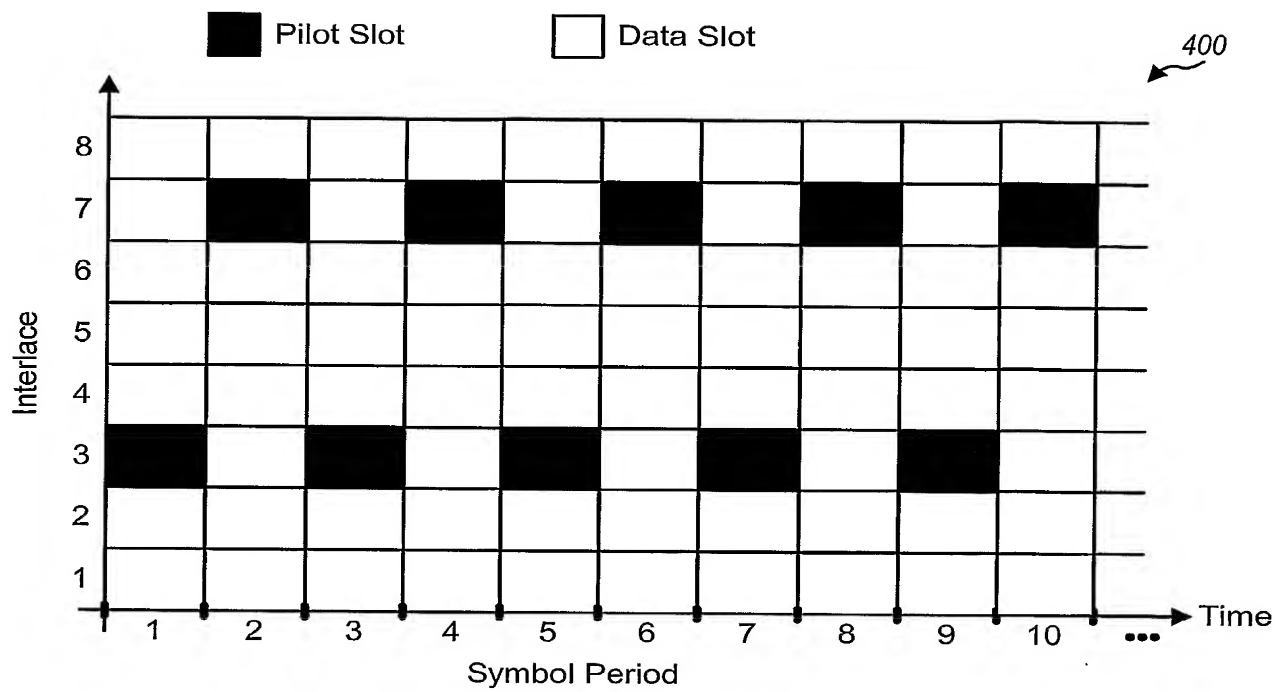

Pilot Slot
Data Slot
400
Interlace
8
7
6
5
4
3
2
1
1
2
3
4
5
6
7
8
9
10
•••
Time
Symbol Period

FIG. 4A

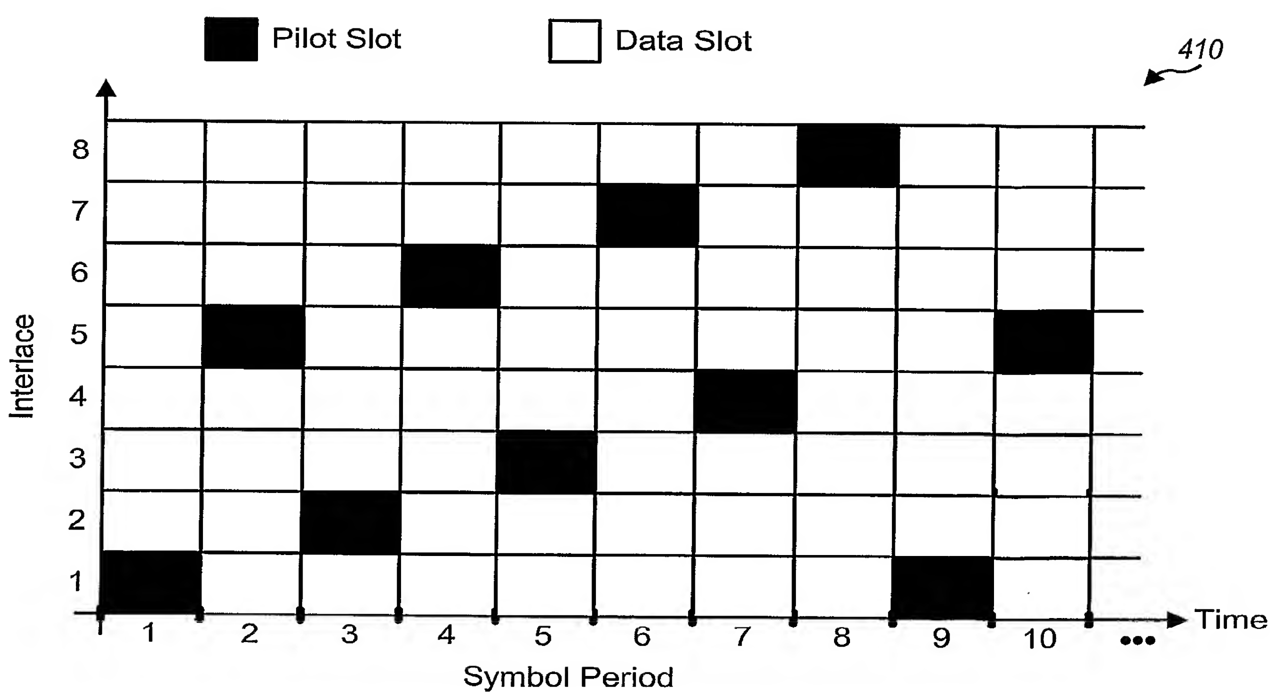

Pilot Slot
Data Slot
410
Interlace
8
7
6
5
4
3
2
1
1
2
3
4
5
6
7
8
9
10
•••
Time
Symbol Period

FIG. 4B

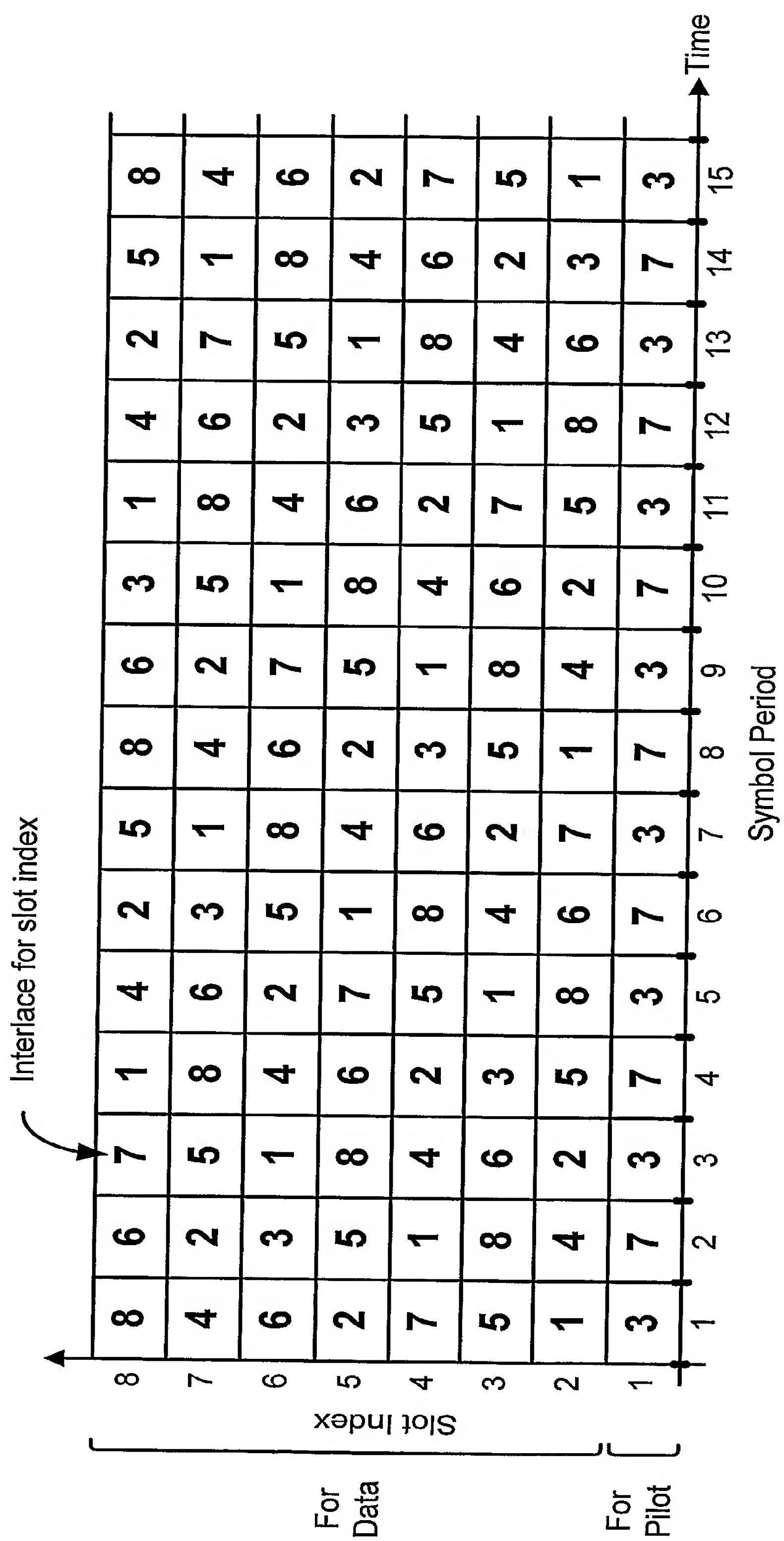
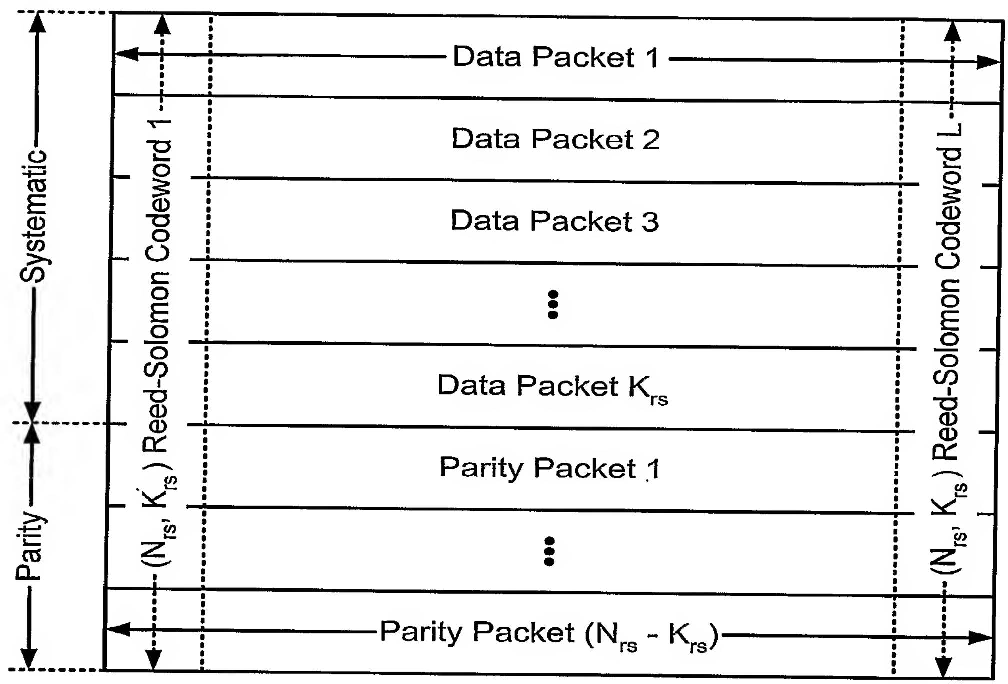
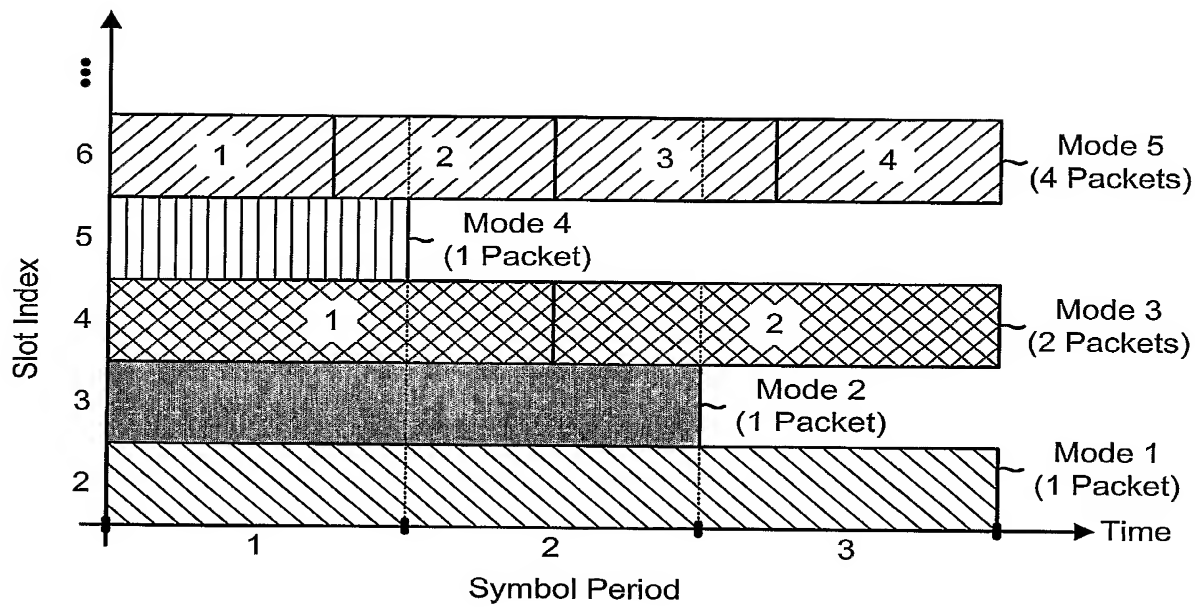

Interlace for slot index
Time
Symbol Period
Slot Index
For Data
For Pilot

FIG. 5



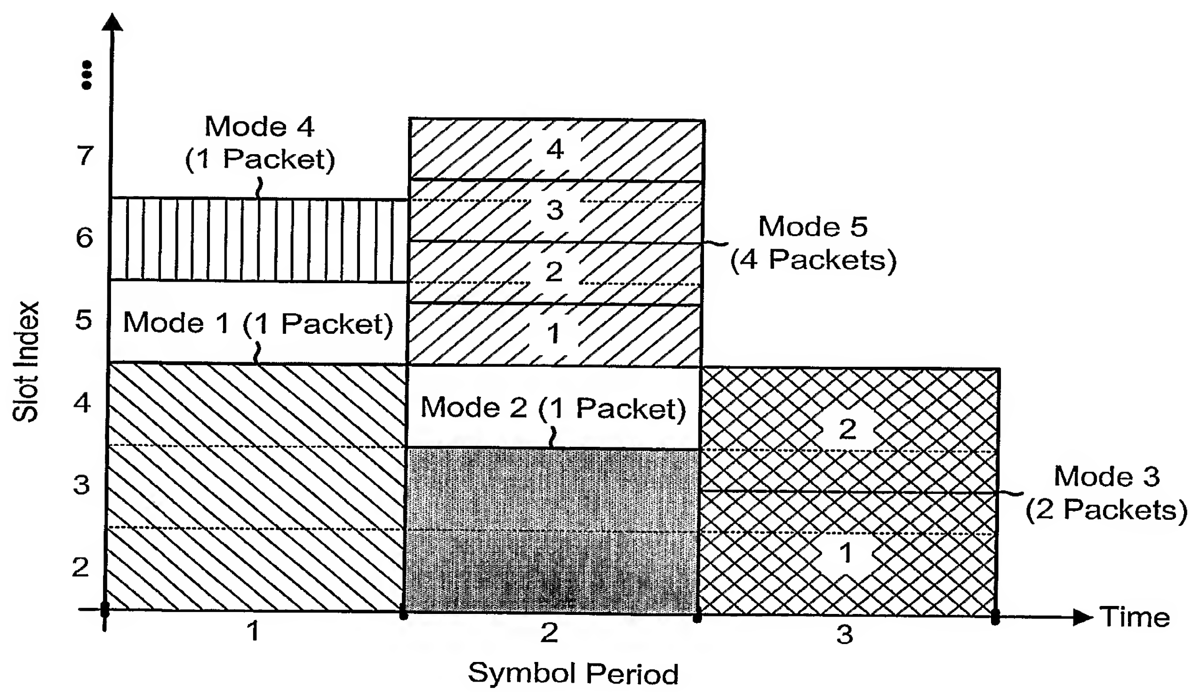
Data Packet 1
Data Packet 2
Data Packet 3
Data Packet K_{rs}
Parity Packet 1
Parity Packet $(N_{rs} - K_{rs})$
Systematic
Parity
(N_{rs}, K_{rs}) Reed-Solomon Codeword 1
(N_{rs}, K_{rs}) Reed-Solomon Codeword L

FIG. 6



Slot Index
6
5
4
3
2
1
2
3
4
Mode 5
(4 Packets)
Mode 4
(1 Packet)
1
2
Mode 3
(2 Packets)
Mode 2
(1 Packet)
Mode 1
(1 Packet)
1
2
3
Time
Symbol Period

FIG. 7A



Slot Index
7
6
5
4
3
2
Mode 4
(1 Packet)
Mode 1 (1 Packet)
4
3
2
1
Mode 5
(4 Packets)
Mode 2 (1 Packet)
2
1
Mode 3
(2 Packets)
1
2
3
Time
Symbol Period

FIG. 7B

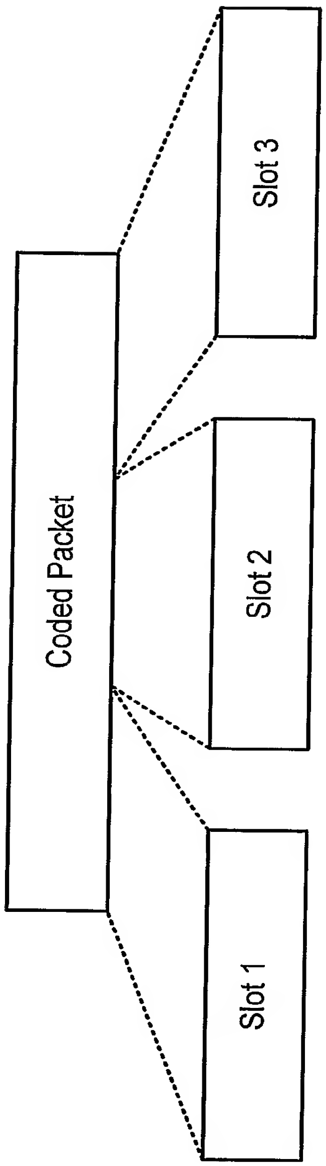

Coded Packet
Slot 1
Slot 2
Slot 3

FIG. 8A

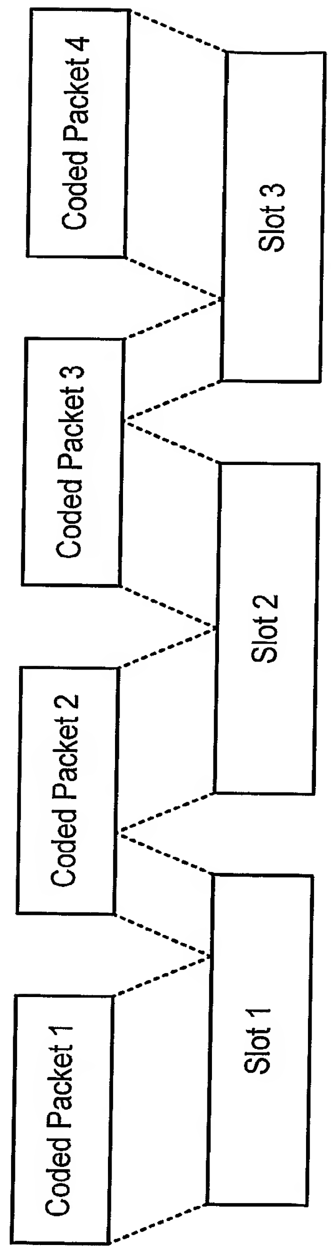

Coded Packet 1
Coded Packet 2
Coded Packet 3
Coded Packet 4
Slot 1
Slot 2
Slot 3

FIG. 8B

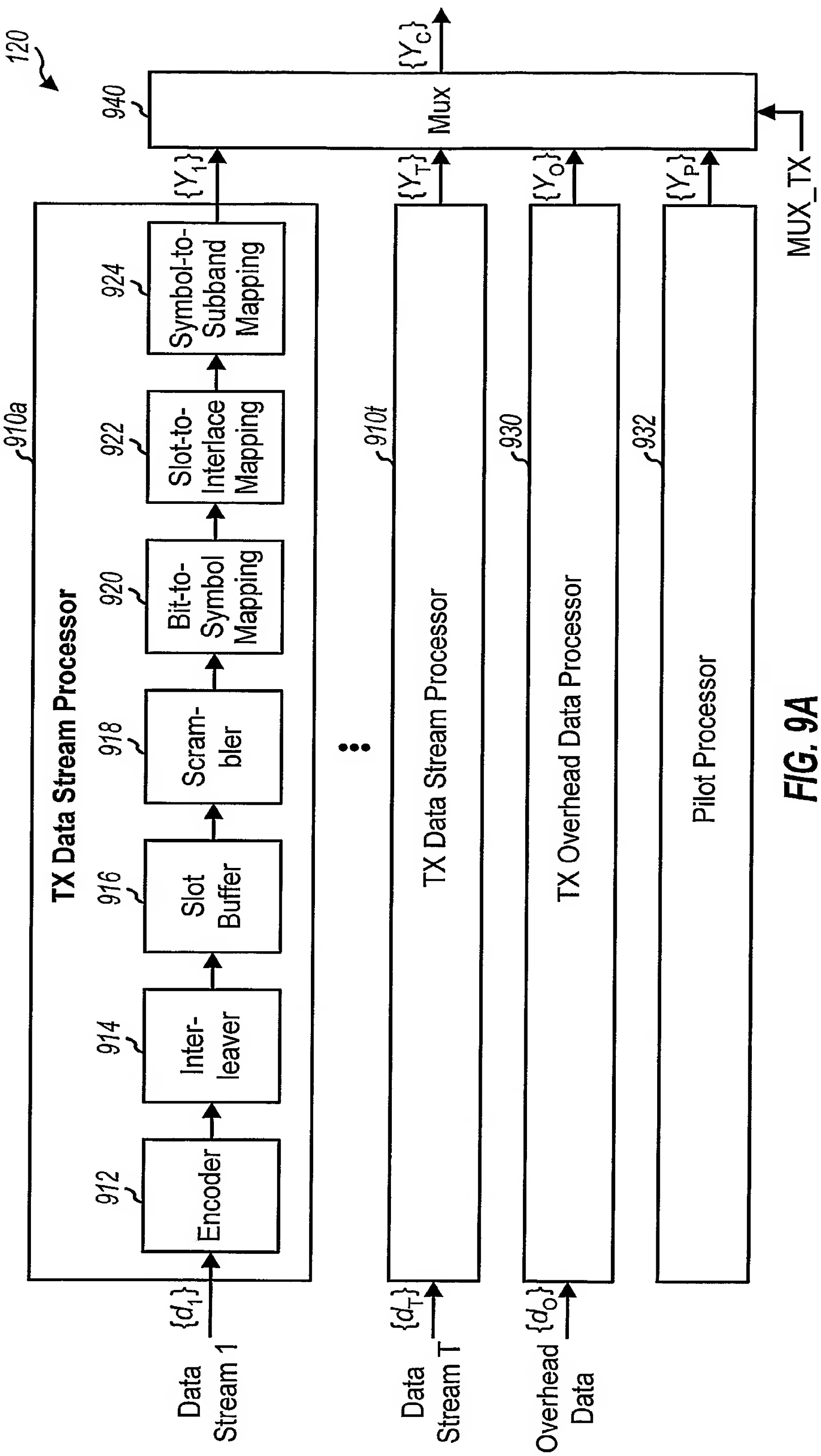

120
TX Data Stream Processor
910a
Data Stream 1
{d1}
912
Encoder
914
Inter-leaver
916
Slot Buffer
918
Scram-bler
920
Bit-to-Symbol Mapping
922
Slot-to-Interlace Mapping
924
Symbol-to-Subband Mapping
{Y1}
940
Mux
{YC}
Data Stream T
{dT}
910t
TX Data Stream Processor
{YT}
Overhead Data
{dO}
930
TX Overhead Data Processor
{YO}
932
Pilot Processor
{YP}
MUX_TX

FIG. 9A

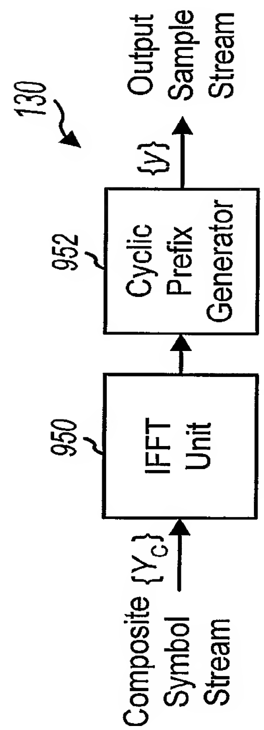

130
Composite Symbol Stream
{YC}
950
IFFT Unit
952
Cyclic Prefix Generator
{y}
Output Sample Stream

FIG. 9B

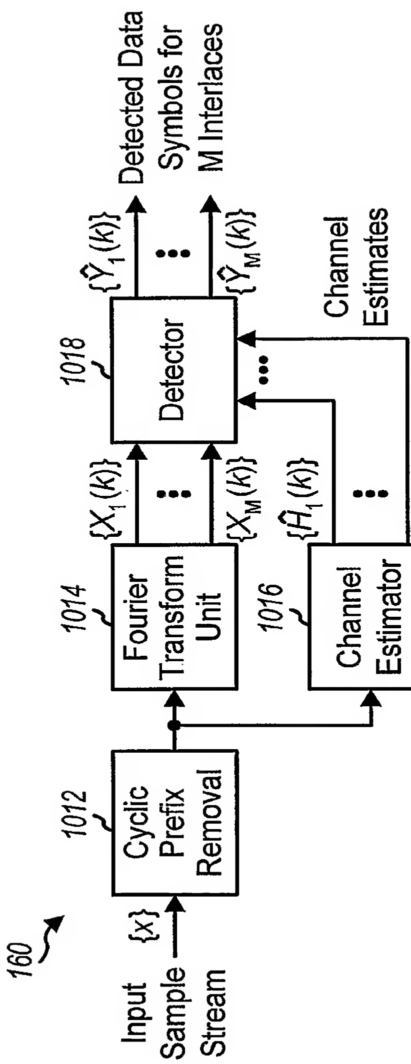


FIG. 10A

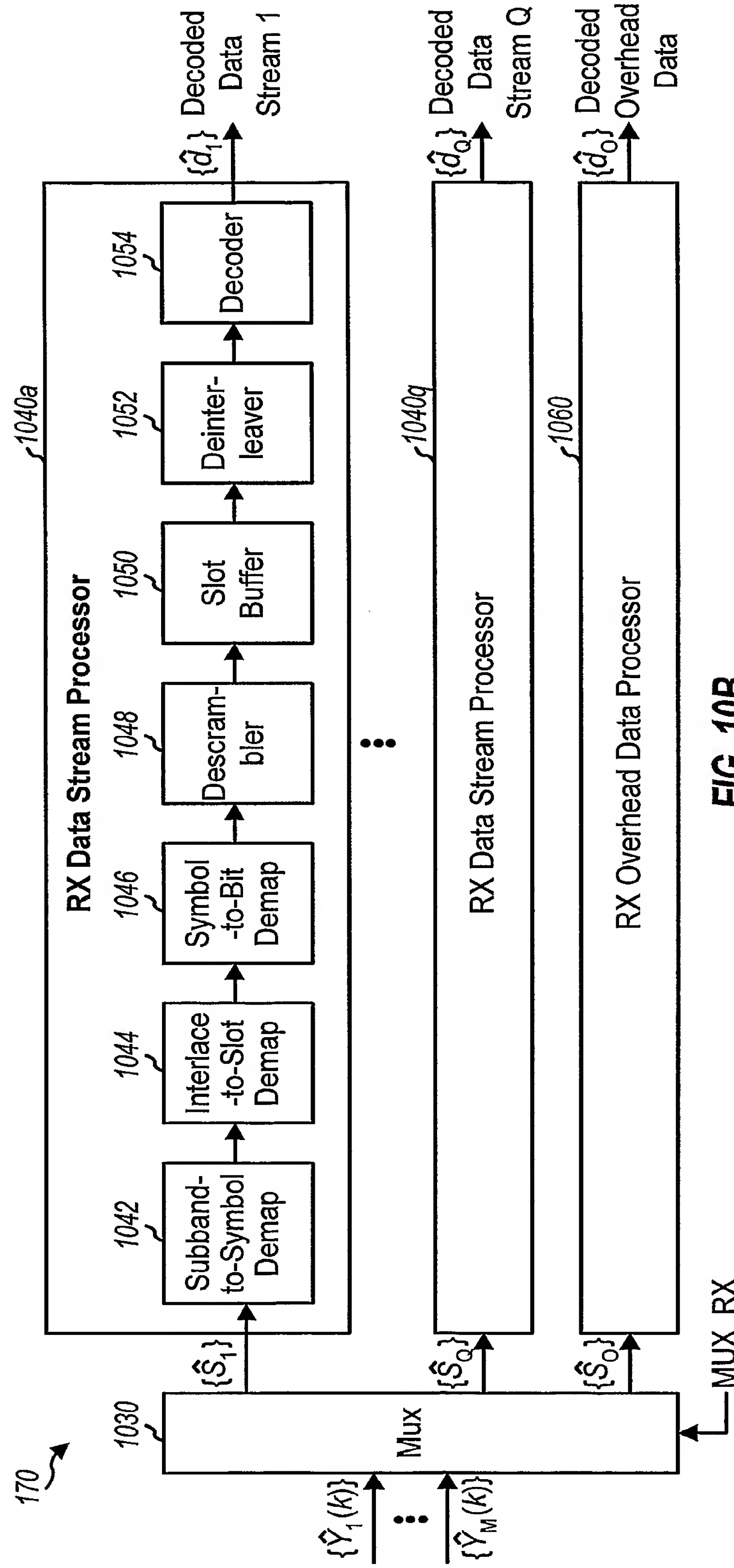


FIG. 10B

INTERNATIONAL SEARCH REPORT

PCT/US2004/035042

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04L27/26 H04L5/02 H04L1/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data, PAJ

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	WO 02/49306 A (BROADSTORM TELECOMMUNICATIONS, INC) 20 June 2002 (2002-06-20) abstract paragraph ‘0002! paragraph ‘0025! - paragraph ‘0028! paragraph ‘0038! - paragraph ‘0053! paragraph ‘0082! - paragraph ‘0088! paragraph ‘0099! paragraph ‘0103! - paragraph ‘0111! -----	1-38,42, 43,45,46
X	WO 02/087104 A (DISENO DE SISTEMAS EN SILICIO, S.A; RIVEIRO INSUA, JUAN, CARLOS; GOMEZ) 31 October 2002 (2002-10-31) abstract -/--	1-6, 11-15, 18-24, 29-32, 36,37, 42,43, 45,46

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

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"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

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"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
15 March 2005	24/03/2005

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL – 2280 HV Rijswijk Tel. (+31–70) 340–2040, Tx. 31 651 epo nl, Fax: (+31–70) 340–3016	Palacián Lisa, M

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INTERNATIONAL SEARCH REPORT

PCT/US2004/035042

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
L,P, X	-& EP 1 388 954 A (DISENO DE SISTEMAS EN SILICIO S.A) 11 February 2004 (2004-02-11) column 1, line 5 - line 30 column 14, line 10 - column 16, line 45 column 19, line 50 - column 20, line 24 column 21, line 49 - column 22, line 22 Family member published later in an official language. Cited for translation purposes. -----	1-6, 11-15, 18-24, 29-32, 36,37, 42,43, 45,46
X	TAKAMURA K ET AL: "FIELD TRIAL RESULTS OF A BAND HOPPING OFDM SYSTEM" VTC 1999-FALL. IEEE VTS 50TH. VEHICULAR TECHNOLOGY CONFERENCE. GATEWAY TO THE 21ST. CENTURY COMMUNICATIONS VILLAGE. AMSTERDAM, SEPT. 19 - 22, 1999, IEEE VEHICULAR TECHNOLGY CONFERENCE, NEW YORK, NY : IEEE, US, vol. VOL. 1 CONF. 50, September 1999 (1999-09), pages 310-314, XP000929061 ISBN: 0-7803-5436-2 I. Introduction II. Band Hopping OFDM system overview Figures 1, 3 abstract -----	1-5,23, 24,29, 31,32, 36,37, 42,43, 45,46
X	WO 02/09455 A (RUNCOM COMMUNICATIONS LTD; HADAD, ZION) 31 January 2002 (2002-01-31) abstract page 4, line 6 - line 17 page 13, line 18 - page 14, line 25 page 17, line 14 - page 25, line 7 -----	1-12, 23-33, 36-38, 42,43, 45,46
X	US 2002/060984 A1 (MICHON VINCENT ET AL) 23 May 2002 (2002-05-23) abstract paragraph ‘0002! paragraph ‘0018! - paragraph ‘0035! paragraph ‘0048! paragraph ‘0052! paragraph ‘0068! - paragraph ‘0094! -----	1-5,7,8, 23,24, 29-47

INTERNATIONAL SEARCH REPORT

PCT/US2004/035042

Patent document cited in search report		Publication date	Patent family member(s)			Publication date
WO 0249306	A	20-06-2002	US	2002119781	A1	29-08-2002
			US	2003169681	A1	11-09-2003
			AU	3258902	A	24-06-2002
			AU	3521702	A	24-06-2002
			AU	3662702	A	24-06-2002
			CA	2431502	A1	20-06-2002
			CA	2431844	A1	20-06-2002
			CA	2431849	A1	20-06-2002
			CN	1481633	A	10-03-2004
			CN	1484906	A	24-03-2004
			EP	1344365	A2	17-09-2003
			JP	2005502218	T	20-01-2005
			JP	2004537875	T	16-12-2004
			JP	2004529524	T	24-09-2004
			WO	0249305	A2	20-06-2002
			WO	0249306	A2	20-06-2002
			WO	0249385	A2	20-06-2002
			US	2002147017	A1	10-10-2002
WO 02087104	A	31-10-2002	ES	2186531	A2	01-05-2003
			BR	0209123	A	15-06-2004
			CA	2444397	A1	31-10-2002
			EP	1388954	A2	11-02-2004
			WO	02087104	A2	31-10-2002
			JP	2004531944	T	14-10-2004
			US	2004136393	A1	15-07-2004
EP 1388954	A	11-02-2004	ES	2186531	A2	01-05-2003
			BR	0209123	A	15-06-2004
			CA	2444397	A1	31-10-2002
			EP	1388954	A2	11-02-2004
			JP	2004531944	T	14-10-2004
			US	2004136393	A1	15-07-2004
			WO	02087104	A2	31-10-2002
WO 0209455	A	31-01-2002	AU	8242301	A	05-02-2002
			WO	0209455	A2	31-01-2002
US 2002060984	A1	23-05-2002	FR	2721461	A1	22-12-1995
			DE	69525624	D1	04-04-2002
			DE	69525624	T2	31-10-2002
			EP	0765549	A1	02-04-1997
			WO	9534967	A1	21-12-1995
			JP	10503894	T	07-04-1998
			US	6282167	B1	28-08-2001

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(19) World Intellectual Property Organization
International Bureau

(43) International Publication Date
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(10) International Publication Number
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(51) International Patent Classification⁷: H04L 27/26, 1/06

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(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:

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10/778,570	13 February 2004 (13.02.2004)	US

(71) Applicant (*for all designated States except US*): **QUALCOMM Incorporated** [US/US]; 5775 Morehouse Drive, San Diego, California 92121 (US).

(72) Inventor; and
(75) Inventor/Applicant (*for US only*): **KADOUS, Tamer** [EG/US]; 5385 Toscana Way, #316, San Diego, California 92122 (US).

(74) Agents: **WADSWORTH, Philip, R.** et al.; 5775 Morehouse Drive, San Diego, California 92121 (US).

(81) Designated States (*unless otherwise indicated, for every kind of national protection available*): AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BW, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MA, MD, MG, MK, MN, MW, MX, MZ, NA, NI, NO, NZ, OM, PG, PH, PL, PT, RO, RU, SC, SD, SE, SG, SK, SL, SY, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, YU, ZA, ZM, ZW.

(84) Designated States (*unless otherwise indicated, for every kind of regional protection available*): ARIPO (BW, GH, GM, KE, LS, MW, MZ, NA, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European (AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HU, IE, IT, LU, MC, NL, PL, PT, RO, SE, SI, SK, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

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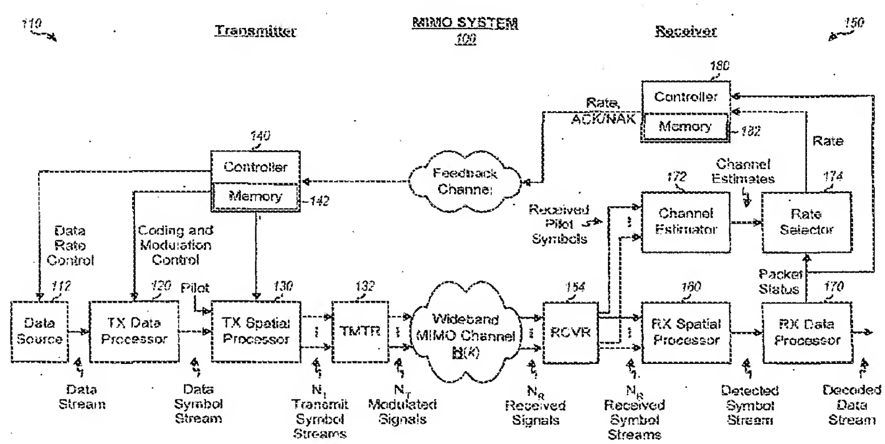
— *as to applicant's entitlement to apply for and be granted a patent (Rule 4.17(ii)) for the following designations AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BW, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DZ, EC, EE,*

[Continued on next page]

(54) Title: RATE SELECTION FOR A MULTI-CARRIER MIMO SYSTEM



(57) Abstract: To select a rate for data transmission in a multi-carrier MIMO system with a multipath MIMO channel, a post-detection SNR for each subband k of each spatial channel is initially determined and used to derive a constrained spectral efficiency based on a constrained spectral efficiency function of SNR and modulation scheme M. An average constrained spectral efficiency for all subbands of all spatial channels used for data transmission is next determined based on the constrained spectral efficiencies for the individual subbands/spatial channels. An equivalent SNR needed by an equivalent system with an AWGN channel to support a data rate of is determined based on an inverse constrained spectral efficiency function . A rate is selected for the multi-carrier MIMO system based on the equivalent SNR. The selected rate is the highest rate among all supported rates with a required SNR less than or equal to the equivalent SNR.

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RATE SELECTION FOR A MULTI-CARRIER MIMO SYSTEM

CROSS-REFERENCE TO RELATED APPLICATION

[0001] This application claims the benefit of U.S. Provisional Patent Application Serial No. 60/514,402, filed October 24, 2003, which is incorporated herein by reference in their entirety.

I. Field

[0002] The present invention relates generally to communication, and more specifically to techniques for performing rate selection for data transmission in a multi-carrier multiple-input multiple-output (MIMO) communication system.

II. Background

[0003] A MIMO system employs multiple (N_T) transmit antennas at a transmitter and multiple (N_R) receive antennas at a receiver for data transmission. A MIMO channel formed by the N_T transmit antennas and N_R receive antennas may be decomposed into N_S spatial channels, where $N_S \le \min\{N_T, N_R\}$. The N_S spatial channels may be used to transmit data in parallel to achieve higher throughput and/or redundantly to achieve greater reliability.

[0004] Orthogonal frequency division multiplexing (OFDM) is a multi-carrier modulation scheme that effectively partitions the overall system bandwidth into multiple (N_F) orthogonal subbands. These subbands are also referred to as tones, subcarriers, bins, and frequency channels. With OFDM, each subband is associated with a respective subcarrier that may be modulated with data.

[0005] For a MIMO system that utilizes OFDM (i.e., a MIMO-OFDM system), N_F subbands are available on each of the N_S spatial channels for data transmission. The N_F subbands of each spatial channel may experience different channel conditions (e.g., different fading, multipath, and interference effects) and may achieve different channel gains and signal-to-noise-and-interference ratios (SNRs). Depending on the multipath profile of the MIMO channel, the channel gains and SNRs may vary widely across the

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N_F subbands of each spatial channel and may further vary widely among the N_S spatial channels.

[0006] For the MIMO-OFDM system, one modulation symbol may be transmitted on each subband of each spatial channel, and up to $N_F \cdot N_S$ modulation symbols may be transmitted simultaneously in each OFDM symbol period. Each transmitted modulation symbol is distorted by the channel gain for the subband of the spatial channel via which the symbol is transmitted and further degraded by channel noise and interference. For a multipath MIMO channel, which is a MIMO channel with a frequency response that is not flat, the number of information bits that may be reliably transmitted on each subband of each spatial channel may vary from subband to subband and from spatial channel to spatial channel. The different transmission capabilities of the different subbands and spatial channels plus the time-variant nature of the MIMO channel make it challenging to ascertain the true transmission capacity of the MIMO-OFDM system.

[0007] There is therefore a need in the art for techniques to accurately determine the transmission capacity of the MIMO-OFDM system for efficient data transmission.

SUMMARY

[0008] Techniques for performing rate selection in a multi-carrier MIMO system (e.g., a MIMO-OFDM system) with a multipath MIMO channel are described herein. In an embodiment, a post-detection SNR, $SNR_\ell(k)$, for each subband k of each spatial channel ℓ used for data transmission is initially determined for a "theoretical" multi-carrier MIMO system that is capable of achieving capacity of the MIMO channel. The post-detection SNR is the SNR after spatial processing or detection at a receiver. The theoretical system has no implementation losses. A constrained spectral efficiency $S_\ell(k)$ for each subband of each spatial channel is then determined based on its post-detection SNR, a modulation scheme M, and a constrained spectral efficiency function $f_{siso}(SNR_\ell(k),\ M)$. An average constrained spectral efficiency S_{avg} for all subbands of all spatial channels used for data transmission is next determined based on the constrained spectral efficiencies for the individual subbands of the spatial channels.

[0009] An equivalent system with an additive white Gaussian noise (AWGN) channel needs an SNR of SNR_{equiv} to achieve a constrained spectral efficiency of S_{avg} with modulation scheme M. An AWGN channel is a channel with a flat frequency response.

The equivalent system also has no implementation losses. The equivalent SNR may be determined based on an inverse constrained spectral efficiency function $f_{siso}^{-1}(S_{avg},\ M)$. A rate R is then selected for data transmission in the multi-carrier MIMO system based on the equivalent SNR. The multi-carrier MIMO system may support a specific set of rates, and the required SNRs for these rates may be determined and stored in a look-up table. The selected rate is the highest rate among the supported rates with a required SNR that is less than or equal to the equivalent SNR. A back-off factor may be computed to account for error in the rate prediction, system losses, and so on. The rate R may then be selected in a manner to account for the back-off factor, as described below.

[0010] Various aspects and embodiments of the invention are described in further detail below.

BRIEF DESCRIPTION OF THE DRAWINGS

[0011] The features and nature of the present invention will become more apparent from the detailed description set forth below when taken in conjunction with the drawings in which like reference characters identify correspondingly throughout and wherein:

[0012] FIG. 1 shows a transmitter and a receiver in a MIMO-OFDM system;

[0013] FIG. 2 illustrates rate selection for the MIMO-OFDM system;

[0014] FIG. 3 shows a process for performing rate selection for a MIMO-OFDM system with a multipath MIMO channel;

[0015] FIG. 4A illustrates constrained spectral efficiencies for N_T spatial channels in the MIMO-OFDM system with the multipath MIMO channel;

[0016] FIG. 4B illustrates constrained spectral efficiency for an equivalent system with an AWGN channel;

[0017] FIG. 5 shows a block diagram of the transmitter;

[0018] FIG. 6 shows a block diagram of the receiver; and

[0019] FIG. 7 shows a receive (RX) spatial processor and an RX data processor that implement iterative detection and decoding (IDD).

DETAILED DESCRIPTION

[0020] The word "exemplary" is used herein to mean "serving as an example, instance, or illustration." Any embodiment or design described herein as "exemplary" is not

necessarily to be construed as preferred or advantageous over other embodiments or designs.

[0021] The rate selection techniques described herein may be used for various types of multi-carrier MIMO system. For clarity, these techniques are specifically described for a MIMO-OFDM system.

[0022] **FIG. 1** shows a block diagram of a transmitter 110 and a receiver 150 in a MIMO-OFDM system 100. At transmitter 110, a transmit (TX) data processor 120 receives packets of data from a data source 112. TX data processor 120 encodes, interleaves, and modulates each data packet in accordance with a rate selected for that packet to obtain a corresponding block of data symbols. As used herein, a data symbol is a modulation symbol for data, and a pilot symbol is a modulation symbol for pilot, which is known *a priori* by both the transmitter and receiver. The selected rate for each data packet may indicate the data rate, coding scheme or code rate, modulation scheme, packet size, and so on for that packet, which are indicated by various controls provided by a controller 140.

[0023] A TX spatial processor 130 receives and spatially processes each data symbol block for transmission on the N_F subbands of the N_T transmit antennas. TX spatial processor 130 further multiplexes in pilot symbols and provides N_T streams of transmit symbols to a transmitter unit (TMTR) 132. Each transmit symbol may be for a data symbol or a pilot symbol. Transmitter unit 132 performs OFDM modulation on the N_T transmit symbol streams to obtain N_T OFDM symbol streams and further processes these OFDM symbol streams to generate N_T modulated signals. Each modulated signal is transmitted from a respective transmit antenna (not shown in FIG. 1) and via a MIMO channel to receiver 150. The MIMO channel distorts the N_T transmitted signals with a MIMO channel response and further degrades the transmitted signals with noise and possibly interference from other transmitters.

[0024] At receiver 150, the N_T transmitted signals are received by each of N_R receive antennas (not shown in FIG. 1), and the N_R received signals from the N_R receive antennas are provided to a receiver unit (RCVR) 154. Receiver unit 154 conditions and digitizes each received signal to obtain a corresponding stream of samples and further performs OFDM demodulation on each sample stream to obtain a stream of received symbols. Receiver unit 154 provides N_R received symbol streams (for data) to an RX spatial processor 160 and received pilot symbols (for pilot) to a channel estimator 172. RX spatial processor 160 spatially processes or detects the N_R received symbol streams

to obtain detected symbols, which are estimates of the data symbols transmitted by transmitter 110.

[0025] An RX data processor 170 receives, demodulates, deinterleaves, and decodes each block of detected symbols in accordance with its selected rate to obtain a corresponding decoded packet, which is an estimate of the data packet sent by transmitter 110. RX data processor 170 also provides the status of each decoded packet, which indicates whether the packet is decoded correctly or in error.

[0026] Channel estimator 172 processes the received pilot symbols and/or received data symbols to obtain channel estimates for the MIMO channel. The channel estimates may include channel gain estimates, SNR estimates, and so on. A rate selector 174 receives the channel estimates and selects a suitable rate for data transmission to receiver 150. A controller 180 receives the selected rate from rate selector 174 and the packet status from RX data processor 170 and assembles feedback information for transmitter 110. The feedback information may include the selected rate, acknowledgments (ACKs) or negative acknowledgments (NAKs) for current and/or prior data packets, and so on. The feedback information is processed and transmitted via a feedback channel to transmitter 110.

[0027] At transmitter 110, the signal(s) transmitted by receiver 150 are received and processed to recover the feedback information sent by receiver 150. Controller 140 receives the recovered feedback information, uses the selected rate to process subsequent data packets to be sent to receiver 150, and uses the ACKs/NAKs to control retransmission of the current and/or prior data packets.

[0028] Controllers 140 and 180 direct the operation at transmitter 110 and receiver 150, respectively. Memory units 142 and 182 provide storage for program codes and data used by controllers 140 and 180, respectively. Memory units 142 and 182 may be internal to controllers 140 and 180, as shown in FIG. 1, or external to these controllers.

[0029] A major challenge for the MIMO-OFDM system is selecting a suitable rate for data transmission based on channel conditions. The goal of the rate selection is to maximize throughput on the N_S spatial channels while meeting certain quality objectives, which may be quantified by a particular packet error rate (e.g., 1% PER).

[0030] The performance of the MIMO-OFDM system is highly dependent on the accuracy of the rate selection. If the selected rate for data transmission is too conservative, then excessive system resources are expended for the data transmission and channel capacity is underutilized. Conversely, if the selected rate is too aggressive, then the receiver may

decode the data transmission in error and system resources may be expended for retransmission. Rate selection for the MIMO-OFDM system is challenging because of the complexity in estimating the true transmission capability of a multipath MIMO channel.

[0031] A multipath MIMO channel formed by the N_T transmit antennas at transmitter 110 and the N_R receive antennas at receiver 150 may be characterized by a set of N_F channel response matrices $\underline{\underline{H}}(k)$, for $k = 1 \ldots N_F$, which may be expressed as:

$$\underline{\underline{H}}(k) = \begin{bmatrix} h_{1,1}(k) & h_{1,2}(k) & \cdots & h_{1,N_T}(k) \\ h_{2,1}(k) & h_{2,2}(k) & \cdots & h_{2,N_T}(k) \\ \vdots & \vdots & \ddots & \vdots \\ h_{N_R,1}(k) & h_{N_R,2}(k) & \cdots & h_{N_R,N_T}(k) \end{bmatrix}, \quad \text{for } k = 1 \ldots N_F, \qquad \text{Eq (1)}$$

where entry $h_{i,j}(k)$, for $i = 1 \ldots N_R$, $j = 1 \ldots N_T$, and $k = 1 \ldots N_F$, denotes the complex channel gain between transmit antenna j and receive antenna i for subband k. For simplicity, the following description assumes that each channel response matrix $\underline{\underline{H}}(k)$ is full rank and the number of spatial channels is $N_S = N_T \le N_R$. In general, a spatial channel is an effective channel between an element of a data symbol vector $\underline{s}(k)$ at the transmitter and a corresponding element of a detected symbol vector $\hat{\underline{s}}(k)$ at the receiver. The vectors $\underline{s}(k)$ and $\hat{\underline{s}}(k)$ are described below. The N_T spatial channels of the MIMO channel are dependent on the spatial processing (if any) performed at the transmitter and the spatial processing performed at the receiver.

[0032] The multipath MIMO channel has a capacity that can be determined in various manners. As used herein, "capacity" denotes the transmission capability of a channel, and "spectral efficiency" denotes the general concept of "capacity per dimension", where the dimension may be frequency and/or space. Spectral efficiency may be given in units of bits per second per Hertz per spatial channel (bps/Hz/ch) for the MIMO-OFDM system. Spectral efficiency is often specified as being either constrained or unconstrained. An "unconstrained" spectral efficiency is typically defined as the theoretical maximum data rate that may be reliably used for a channel with a given channel response and noise variance. A "constrained" spectral efficiency is further dependent on the specific modulation scheme used for data transmission. The constrained capacity (due to the fact that modulation symbols are restricted to specific

points on a signal constellation) is lower than the unconstrained capacity (which is not confined by any signal constellation).

[0033] **FIG. 2** graphically illustrates a technique for performing rate selection for a MIMO-OFDM system with a multipath MIMO channel. For a given multipath MIMO channel defined by a channel response of $\underline{\mathbf{H}}(k)$, for $k = 1 \ldots N_F$, and a noise variance of N_0, a theoretical MIMO-OFDM system has an average constrained spectral efficiency of S_{avg} with modulation scheme M. As used herein, a "theoretical" system is one without any losses, and a "practical" system is one with implementation losses (e.g., due to hardware imperfections), code loss due to the fact that practical codes do not work at capacity, and any other losses. The theoretical and practical systems both use one or more modulation schemes for data transmission and are defined by constrained spectral efficiencies. The average constrained spectral efficiency S_{avg} may be determined as described below. In general, different modulation schemes may be used for different subbands and/or spatial channels. For simplicity, the following description assumes that the same modulation scheme M is used for all subbands of all spatial channels available for data transmission.

[0034] An equivalent system with an AWGN channel needs an SNR of SNR_{equiv} to achieve a constrained spectral efficiency of S_{avg} with modulation scheme M. This equivalent system also has no losses. The equivalent SNR may be derived as described below.

[0035] A practical MIMO-OFDM system with an AWGN channel requires an SNR of SNR_{req} or better to support rate R, which is associated with modulation scheme M, coding scheme C, and data rate D. The data rate D is given in units of bps/Hertz/ch, which is the same unit used for spectral efficiency. The rate R may be selected as the highest rate supported by the system with a required SNR equal to or less than the equivalent SNR, as described below. The required SNR is dependent on modulation scheme M, coding scheme C, and other system losses. The required SNR may be determined for each supported rate (e.g., based on computer simulation, empirical measurement, or some other means) and stored in a look-up table.

[0036] A practical MIMO-OFDM system with a multipath MIMO channel (e.g., MIMO-OFDM system 100) is deemed to support rate R with modulation scheme M and coding scheme C if the required SNR is less than or equal to the equivalent SNR. As

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the rate increases, the required SNR increases for the practical system while the equivalent SNR is approximately constant since it is defined by the channel response $\underline{\mathbf{H}}(k)$ and the noise variance N_0. The maximum rate that may be supported by the practical MIMO-OFDM system with the multipath MIMO channel is thus limited by the channel conditions. Details of the rate selection are described below.

[0037] An ideal system having an unconstrained spectral efficiency can be analyzed and used for rate selection for the practical system having a constrained spectral efficiency. An unconstrained spectral efficiency for each subband of the multipath MIMO channel may be determined based on an unconstrained MIMO spectral efficiency function, as follows:

$$S_{unconst}(k) = \frac{1}{\mathrm{N_T}} \cdot \log_2 \left[\det \left(\underline{\mathbf{I}} + \underline{\mathbf{H}}(k) \cdot \underline{\Gamma}(k) \cdot \underline{\mathbf{H}}^H(k) \right) \right], \quad \text{for } k = 1 \ \dots \ \mathrm{N_F}, \qquad \text{Eq (2)}$$

where $\det(\underline{\mathbf{M}})$ denotes the determinant of $\underline{\mathbf{M}}$, $\underline{\mathbf{I}}$ is the identity matrix, $S_{unconst}(k)$ is the unconstrained spectral efficiency of $\underline{\mathbf{H}}(k)$, $\underline{\Gamma}(k)$ is a matrix that determines the power used for the transmit antennas, and "^{H}" denotes a conjugate transpose. If the channel response $\underline{\mathbf{H}}(k)$ is only known by the receiver, then $\underline{\Gamma}(k)$ is equal to the identity matrix (i.e., $\underline{\Gamma}(k) = \underline{\mathbf{I}}$).

[0038] For a capacity achieving MIMO-OFDM system, which is a system that can transmit and receive data at the capacity of the MIMO channel assuming that a capacity achieving code is available for use, the unconstrained spectral efficiency for each subband of the MIMO channel may be determined based on an unconstrained SISO spectral efficiency function, as follows:

$$S_{unconst}(k) = \frac{1}{\mathrm{N_T}} \cdot \sum_{\ell=1}^{\mathrm{N_T}} \log_2 \left[1 + \mathrm{SNR}_{\ell}(k) \right], \quad \text{for } k = 1 \ \dots \ \mathrm{N_F}, \qquad \text{Eq (3)}$$

where $\mathrm{SNR}_{\ell}(k)$ is the post-detection SNR for subband k of spatial channel ℓ for the capacity achieving system. The post-detection SNR is the SNR achieved for a detected symbol stream after the receiver spatial processing to remove interference from the other symbol streams. The post-detection SNR in equation (3) may be obtained, for example, by a receiver that uses a successive interference cancellation (SIC) technique with a minimum mean square error (MMSE) detector, as described below. Equations

(2) and (3) indicate that, for the capacity achieving system, the unconstrained spectral efficiency of the MIMO channel is equal to the sum of the unconstrained spectral efficiencies of the N_T single-input single-output (SISO) channels that make up the MIMO channel. Each SISO channel corresponds to a spatial channel of the MIMO channel.

[0039] If a single data rate is used for data transmission on all N_F subbands of all N_T transmit antennas, then this single data rate may be set to the average unconstrained spectral efficiency for the N_F subbands of the MIMO channel, as follows:

$$D_{unconst} = \frac{1}{N_F} \cdot \sum_{k=1}^{N_F} S_{unconst}(k) . \qquad \text{Eq (4)}$$

Substituting the unconstrained SISO spectral efficiency function in equation (3) into equation (4), the single data rate may be expressed as:

$$D_{unconst} = \frac{1}{N_F N_T} \cdot \sum_{k=1}^{N_F} \sum_{\ell=1}^{N_T} \log_2 \left[1 + \mathrm{SNR}_\ell(k) \right] . \qquad \text{Eq (5)}$$

[0040] The data rate $D_{unconst}$ is obtained based on the average unconstrained spectral efficiency and is suitable for the ideal MIMO-OFDM system, which is not restricted to a specific modulation scheme. The practical MIMO-OFDM system uses one or more specific modulation schemes for data transmission and has a constrained spectral efficiency that is less than the unconstrained capacity. The data rate $D_{unconst}$ derived based on equation (5) is an optimistic data rate for the practical MIMO-OFDM system. A more accurate data rate may be obtained for the practical MIMO-OFDM system based on a constrained capacity function, instead of an unconstrained capacity function, as described below.

[0041] **FIG. 3** shows a process 300 for performing rate selection for a practical MIMO-OFDM system with a multipath MIMO channel. Process 300 may be performed by rate selector 174 or some other processing unit at the receiver. Initially, an average constrained spectral efficiency S_{avg} for the MIMO channel is determined (block 310). This may be achieved in several ways.

[0042] If a constrained MIMO spectral efficiency function $f_{mimo}(\underline{\mathbf{H}}(k),\ M)$ is available, then the constrained spectral efficiency for each subband of the MIMO channel may be computed based on this function (block 312), as follows:

$$S_{mimo}(k) = \frac{1}{\mathrm{N_T}} \cdot f_{mimo}(\underline{\mathbf{H}}(k),\ M)\ , \quad \text{for } k = 1\ \ldots\ \mathrm{N_F}\,. \qquad \text{Eq (6)}$$

The average constrained spectral efficiency S_{avg} for all subbands of the MIMO channel may then be computed (block 314), as follows:

$$S_{avg} = \frac{1}{\mathrm{N_F}} \cdot \sum_{k=1}^{\mathrm{N_F}} S_{mimo}(k)\ . \qquad \text{Eq (7)}$$

[0043] The constrained MIMO spectral efficiency function $f_{mimo}(\underline{\mathbf{H}}(k),\ M)$ is likely to be a complex equation with no closed form solution or may not even be available. In this case, the MIMO channel may be decomposed into $\mathrm{N_T}$ SISO channels, and the average constrained spectral efficiency S_{avg} for the MIMO channel may be determined based on the constrained spectral efficiencies of the individual SISO channels. Since the unconstrained spectral efficiency of the MIMO channel is equal to the sum of the unconstrained spectral efficiencies of the $\mathrm{N_T}$ SISO channels for a capacity achieving system, as described above, the constrained spectral efficiency of the MIMO channel can be assumed to be equal to the sum of the constrained spectral efficiencies of the $\mathrm{N_T}$ SISO channels for the capacity achieving system.

[0044] To compute S_{avg}, the post-detection SNR $SNR_\ell(k)$ for each subband k of each spatial channel ℓ may be determined for the capacity achieving system, as described below (block 322). The constrained spectral efficiency $S_\ell(k)$ for each subband of each spatial channel is then determined based on a constrained SISO spectral efficiency function $f_{siso}(SNR_\ell(k),\ M)$ (block 324), as follows:

$$S_\ell(k) = f_{siso}(SNR_\ell(k),\ M)\ , \quad \text{for } k = 1\ \ldots\ \mathrm{N_F} \text{ and } \ell = 1\ \ldots\ \mathrm{N_T}\,. \qquad \text{Eq (8)}$$

The constrained SISO spectral efficiency function $f_{siso}(SNR_\ell(k),\ M)$ may be defined as:

$$f_{siso}(SNR_\ell(k),\ M) =$$

$$B - \frac{1}{2^B}\sum_{i=1}^{2^B} E\left[\log_2 \sum_{j=1}^{2^B} \exp\left(-\mathrm{SNR}_\ell(k)\cdot(|a_i - a_j|^2 + 2\,\mathrm{Re}\{\eta^*(a_i - a_j)\})\right)\right], \qquad \text{Eq (9)}$$

where B is the number of bits for each modulation symbol for modulation scheme M;

a_i and a_j are signal points in the 2^B-ary constellation for modulation scheme M;

η is a complex Gaussian random variable with zero mean and a variance of $1/\mathrm{SNR}_\ell(k)$; and

$E[\cdot]$ is an expectation operation taken with respect to η in equation (9).

Modulation scheme M is associated with a 2^B-ary constellation (e.g., 2^B-ary QAM) that contains 2^B signal points. Each signal point in the constellation is labeled with a different B-bit value.

[0045] The constrained SISO spectral efficiency function shown in equation (9) does not have a closed form solution. This function may be numerically solved for various SNR values for each modulation scheme, and the results may be stored in a look-up table. Thereafter, the constrained SISO spectral efficiency function may be evaluated by accessing the look-up table with the modulation scheme M and the post-detection SNR $SNR_\ell(k)$.

[0046] The average constrained spectral efficiency S_{avg} for all subbands of all spatial channels may then be computed (block 326), as follows:

$$S_{avg} = \frac{1}{\mathrm{N_F}\cdot\mathrm{N_T}}\cdot\sum_{k=1}^{\mathrm{N_F}}\sum_{\ell=1}^{\mathrm{N_T}} S_\ell(k)\ . \qquad \text{Eq (10)}$$

[0047] The average constrained spectral efficiency S_{avg} may be computed for a practical MIMO-OFDM system with a multipath MIMO channel in various manners. Two exemplary methods are described above. Other methods may also be used.

[0048] An equivalent system with an AWGN channel would require an SNR of SNR_{equiv} to achieve a constrained spectral efficiency of S_{avg} with modulation scheme M. The equivalent SNR may be determined based on an inverse constrained SISO spectral

efficiency function $f_{siso}^{-1}(S_{avg}, M)$ (block 330). The constrained SISO spectral efficiency function $f_{siso}(x)$ takes two inputs, $SNR_\ell(k)$ and M, and maps them to a constrained spectral efficiency $S_\ell(k)$. Here, x represents the set of pertinent variables for the function. The inverse constrained SISO spectral efficiency function $f_{siso}^{-1}(x)$ takes two inputs, S_{avg} and M, and maps them to an SNR value, as follows:

$$SNR_{equiv} = f_{siso}^{-1}(S_{avg}, M) \; . \qquad \text{Eq (11)}$$

The inverse function $f_{siso}^{-1}(S_{avg}, M)$ may be determined once for each supported modulation scheme and stored in a look-up table.

[0049] The highest rate that may be used for data transmission in a practical MIMO-OFDM system with an AWGN channel is then determined based on the equivalent SNR for the equivalent system (block 332). The practical MIMO-OFDM system may support a set of P rates, $R = \{R(m), \; m = 1, \; 2, \; \ldots \; \text{P}\}$, where m is a rate index. Only the P rates in set R are available for use for data transmission. Each rate $R(m)$ in set R may be associated with a specific modulation scheme $M(m)$, a specific code rate or coding scheme $C(m)$, a specific data rate $D(m)$, and a specific required SNR $SNR_{req}(m)$, as follows:

$$R(m) \leftrightarrow [M(m), \; C(m), \; D(m), \; SNR_{req}(m)] \; , \quad \text{for } m = 1 \; \ldots \; \text{P} . \qquad \text{Eq (12)}$$

For each rate $R(m)$, the data rate $D(m)$ is determined by the modulation scheme $M(m)$ and the code rate $C(m)$. For example, a rate associated with a modulation scheme of QPSK (with two bits per modulation symbol) and a code rate of 1/2 would have a data rate of 1.0 information bit per modulation symbol. Expression (12) states that data rate $D(m)$ may be transmitted using modulation scheme $M(m)$ and code rate $C(m)$ and further requires an SNR of $SNR_{req}(m)$ or better to achieve a PER of P_e. The required SNR accounts for system losses in the practical system and may be determined by computer simulation, empirical measurements, and so on. The set of supported rates and their required SNRs may be stored in a look-up table. The equivalent SNR SNR_{equiv} may be provided to the look-up table, which then returns the rate $R = R(m_s)$ associated

with the highest data rate supported by SNR_{equiv}. The selected rate R is such that the following conditions are met: (1) modulation scheme M is used for data transmission, or $M(m_s) = M$, (2) the required SNR is less than or equal to the equivalent SNR, or $SNR_{req}(m_s) \leq SNR_{equiv}$, and (3) the maximum data rate is selected, or $D_s = \max_m \{D(m)\}$, subject to the other conditions. The selected rate R includes a back-off factor that accounts for loss due to the selected code rate $C(m_s)$, which may not be able to achieve capacity. This back off occurs in condition (2) above.

[0050] The data rate D_s is indicative of the maximum data rate that can be transmitted on each subband of each spatial channel for a capacity achieving system. An aggregate data rate for all N_T spatial channels may be computed, as follows:

$$D_{total} = D_s \cdot N_T \ . \qquad \text{Eq (13)}$$

The aggregate data rate is given in units of bps/Hz, which is normalized to frequency. The factor of N_F is thus not included in equation (13). The aggregate data rate represents a prediction of the data rate that can be supported by the practical MIMO-OFDM system with the multipath MIMO channel for the desired PER of P_e.

[0051] The rate selection technique described above assumes that the practical MIMO-OFDM system is capable of achieving capacity with modulation scheme M. Several transmission schemes that can achieve capacity are described below. The selected rate R may be an accurate rate for such a system and may be used for data transmission without any modification.

[0052] However, as with any rate prediction scheme, there will inevitably be errors in the rate prediction. Moreover, the practical system may not be able to achieve capacity and/or may have other losses that are unaccounted for by the selected rate R. In this case, to ensure that the desired PER can be achieved, errors in the rate prediction may be estimated and an additional back-off factor may be derived. The rate obtained in block 332 may then be reduced by the additional back-off factor to obtain a final rate for data transmission via the multipath MIMO channel. Alternatively, the average constrained spectral efficiency S_{avg} may be reduced by the additional back-off factor, and the reduced average constrained spectral efficiency may be provided to the look-up table to obtain the rate for data transmission. In any case, the additional back-off factor

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reduces the throughput of the system. Thus, it is desirable to keep this back-off factor as small as possible while still achieving the desired PER. An accurate rate prediction scheme, such as the one described herein, may minimize the amount of additional back off to apply and hence maximize system capacity.

[0053] The rate selection described above may be performed continually for each time interval, which may be of any duration (e.g., one OFDM symbol period). It is desirable to use the selected rate for data transmission as soon as possible to minimize the amount of time between the selection of the rate and the use of the rate.

[0054] **FIG. 4A** illustrates the constrained spectral efficiencies for the N_T spatial channels in the MIMO-OFDM system with the multipath MIMO channel. For each spatial channel, a plot 410 of the constrained spectral efficiencies for the N_F subbands may be derived based on the post-detection SNRs, the modulation scheme M, and the constrained SISO spectral efficiency function $f_{siso}(SNR_\ell(k),\ M)$, as shown in equations (8) and (9). Plots 410a through 410t for the N_T spatial channels may be different because of different fading for these spatial channels, as shown in FIG. 4A.

[0055] **FIG. 4B** illustrates the constrained spectral efficiency of the equivalent system with the AWGN channel. A plot 420 is formed by concatenation of plots 410a through 410t for the N_T spatial channels in FIG. 4A. A plot 422 shows the constrained spectral efficiency for the equivalent system, which is the average of the constrained spectral efficiencies for plots 410a through 410t.

[0056] The rate selection described above includes a back-off factor for code loss but otherwise assumes that the MIMO-OFDM system can achieve capacity. Two exemplary transmission schemes capable of achieving capacity are described below.

[0057] In a first transmission scheme, the transmitter transmits data on "eigenmodes" of the MIMO channel. The eigenmodes may be viewed as orthogonal spatial channels obtained by decomposing the MIMO channel. The channel response matrix $\underline{\mathbf{H}}(k)$ for each subband may be decomposed using eigenvalue decomposition, as follows:

$$\underline{\mathbf{R}}(k) = \underline{\mathbf{H}}^H(k) \cdot \underline{\mathbf{H}}(k) = \underline{\mathbf{E}}(k) \cdot \underline{\Lambda}(k) \cdot \underline{\mathbf{E}}^H(k)\ , \quad \text{for } k = 1 \ \ldots \ N_F, \qquad \text{Eq (14)}$$

where $\underline{\mathbf{R}}(k)$ is an $N_T \times N_T$ correlation matrix of $\underline{\mathbf{H}}(k)$;

$\underline{\mathbf{E}}(k)$ is an $N_T \times N_T$ unitary matrix whose columns are eigenvectors of $\underline{\mathbf{R}}(k)$; and

$\underline{\Lambda}(k)$ is an $N_T \times N_T$ diagonal matrix of eigenvalues of $\underline{\mathbf{R}}(k)$.

A unitary matrix $\underline{\mathbf{U}}$ is characterized by the property $\underline{\mathbf{U}}^H \cdot \underline{\mathbf{U}} = \underline{\mathbf{I}}$. The columns of a unitary matrix are orthogonal to one another.

[0058] The transmitter performs spatial processing as follows:

$$\underline{\mathbf{x}}(k) = \underline{\mathbf{E}}(k) \cdot \underline{\mathbf{s}}(k) \ , \quad \text{for } k = 1 \ \ldots \ N_F, \qquad \text{Eq (15)}$$

where $\underline{\mathbf{s}}(k)$ is an $N_T \times 1$ vector with N_T data symbols to be sent on the N_T eigenmodes of subband k; and

$\underline{\mathbf{x}}(k)$ is an $N_T \times 1$ vector with N_T transmit symbols to be transmitted from the N_T transmit antennas on subband k.

[0059] The received symbols at the receiver may be expressed as:

$$\underline{\mathbf{r}}_{em}(k) = \underline{\mathbf{H}}(k) \cdot \underline{\mathbf{x}}(k) + \underline{\mathbf{n}}(k) \ , \quad \text{for } k = 1 \ \ldots \ N_F, \qquad \text{Eq (16)}$$

where $\underline{\mathbf{r}}_{em}(k)$ is an $N_R \times 1$ vector with N_R received symbols obtained via the N_R receive antennas on subband k; and

$\underline{\mathbf{n}}$ is an $N_R \times 1$ vector of noise and interference for subband k.

The noise vector $\underline{\mathbf{n}}(k)$ is assumed to have zero mean and a covariance matrix of $\underline{\mathbf{\Lambda}}_n(k) = N_0 \cdot \underline{\mathbf{I}}$, where N_0 is the noise variance.

[0060] The receiver performs receiver spatial processing/detection, as follows:

$$\hat{\underline{\mathbf{s}}}_{em}(k) = \underline{\mathbf{\Lambda}}^{-1}(k) \cdot \underline{\mathbf{E}}^H(k) \cdot \underline{\mathbf{H}}^H(k) \cdot \underline{\mathbf{r}}_{em}(k) = \underline{\mathbf{s}}(k) + \underline{\mathbf{n}}_{em}(k) \ , \quad k = 1 \ \ldots \ N_F, \qquad \text{Eq (17)}$$

where $\hat{\underline{\mathbf{s}}}_{em}(k)$ is an $N_T \times 1$ vector with N_T detected symbols for subband k, which are estimates of the N_T data symbols in $\underline{\mathbf{s}}(k)$; and

$\underline{\mathbf{n}}_{em}(k) = \underline{\mathbf{\Lambda}}^{-1}(k) \cdot \underline{\mathbf{E}}^H(k) \cdot \underline{\mathbf{H}}^H(k) \cdot \underline{\mathbf{n}}(k)$ is the post-detection interference and noise after the spatial processing at the receiver.

Each eigenmode is an effective channel between an element of the data symbol vector $\underline{\mathbf{s}}(k)$ and a corresponding element of the detected symbol vector $\hat{\underline{\mathbf{s}}}_{em}(k)$.

[0061] The SNR for each subband of each eigenmode may be expressed as:

$$SNR_{em,\ell}(k) = \frac{P_\ell(k) \cdot \lambda_\ell(k)}{N_0} \ , \quad \text{for } k = 1 \ \ldots \ N_F \text{ and } \ell = 1 \ \ldots \ N_T, \qquad \text{Eq (18)}$$

where $P_\ell(k)$ is the transmit power used for eigenmode ℓ of subband k;

$\lambda_\ell(k)$ is the eigenvalue for eigenmode ℓ of subband k, which is the ℓ-th diagonal element of $\underline{\Lambda}(k)$; and

$SNR_{em,\ell}(k)$ is the post-detection SNR for eigenmode ℓ of subband k.

[0062] In a second transmission scheme, the transmitter encodes and modulates data to obtain data symbols, demultiplexes the data symbols into N_T data symbol streams, and transmits the N_T data symbol streams simultaneously from the N_T transmit antennas. The received symbols at the receiver may be expressed as:

$$\underline{r}_{am}(k) = \underline{\mathbf{H}}(k) \cdot \underline{s}(k) + \underline{n}(k) \ , \quad \text{for } k = 1 \ \dots \ N_F . \qquad \text{Eq (19)}$$

[0063] The receiver performs receiver spatial processing/detection on the N_R received symbols for each subband to recover the N_T data symbols transmitted on that subband. The receiver spatial processing may be performed with a minimum mean square error (MMSE) detector, a maximal ratio combining (MRC) detector, a linear zero-forcing (ZF) detector, an MMSE linear equalizer (MMSE-LE), a decision feedback equalizer (DFE), or some other detector/equalizer.

[0064] The receiver may also process the N_R received symbol streams using a successive interference cancellation (SIC) technique to recover the N_T data symbol streams. The SIC technique may be used when the transmitter independently processes the N_T data symbol streams so that the receiver can individually recover each data symbol stream. The receiver recovers the N_T data symbol streams in N_T successive stages, one data symbol stream in each stage.

[0065] For the first stage, the receiver initially performs receiver spatial processing/detection on the N_R received symbol streams (e.g., using an MMSE, MRC, or zero-forcing detector) and obtains one detected symbol stream. The receiver further demodulates, deinterleaves, and decodes the detected symbol stream to obtain a decoded data stream. The receiver then estimates the interference this decoded data stream causes to the other $N_T - 1$ data symbol streams not yet recovered, cancels the estimated interference from the N_R received symbol streams, and obtains N_R modified symbol streams for the next stage. The receiver then repeats the same processing on the N_R modified symbol streams to recover another data symbol stream. For simplicity, the following description assumes that the N_T data symbol streams are recovered in

sequential order, i.e., data symbol stream $\{s_\ell(k)\}$ sent from transmit antenna ℓ is recovered in the ℓ-th stage, for $\ell = 1 \ldots N_T$.

[0066] For a SIC with MMSE receiver, an MMSE detector is derived for each subband of stage ℓ, for $\ell = 1 \ldots N_T$, as follows:

$$\underline{\mathbf{W}}_{mmse,\ell}(k) = (\underline{\mathbf{H}}_\ell(k) \cdot \underline{\mathbf{H}}_\ell^H(k) + N_0 \cdot \underline{\mathbf{I}})^{-1} \cdot \underline{\mathbf{H}}_\ell(k) \; , \quad \text{for } k = 1 \ldots N_F \, , \qquad \text{Eq (20)}$$

where $\underline{\mathbf{W}}_{mmse,\ell}(k)$ is an $N_R \times (N_T - \ell + 1)$ matrix for the MMSE detector for subband k in stage ℓ; and

$\underline{\mathbf{H}}_\ell(k)$ is an $N_R \times (N_T - \ell + 1)$ reduced channel response matrix for subband k in stage ℓ.

The reduced channel response matrix $\underline{\mathbf{H}}_\ell(k)$ is obtained by removing $\ell - 1$ columns in the original matrix $\underline{\mathbf{H}}(k)$ corresponding to the $\ell - 1$ data symbol streams already recovered in the $\ell - 1$ prior stages.

[0067] The receiver performs detection for each subband in stage ℓ, as follows:

$$\hat{s}_{mmse,\ell}(k) = \underline{\mathbf{w}}_{mmse,\ell}^H(k) \cdot \underline{\mathbf{r}}_\ell(k) = s_\ell(k) + \underline{\mathbf{w}}_{mmse,\ell}^H(k) \cdot \underline{\mathbf{n}}_\ell(k) \; , \qquad \text{Eq (21)}$$

where $\underline{\mathbf{w}}_{mmse,\ell}(k)$ is a column of $\underline{\mathbf{W}}_{mmse,\ell}(k)$ corresponding to transmit antenna ℓ;

$\hat{s}_{mmse,\ell}(k)$ is the MMSE detected symbol for subband k in stage ℓ; and

$\underline{\mathbf{w}}_{mmse,\ell}^H(k) \cdot \underline{\mathbf{n}}_\ell(k)$ is the post-detection noise for the detected symbol $\hat{s}_{mmse,\ell}(k)$.

[0068] The SNR for each subband of each transmit antenna may be expressed as:

$$SNR_{mmse,\ell}(k) = \frac{P_\ell(k)}{N_0 \cdot \| \underline{\mathbf{w}}_{mmse,\ell}(k) \|^2} \; , \qquad \text{Eq (22)}$$

where $N_0 \cdot \| \underline{\mathbf{w}}_{mmse,\ell}(k) \|^2$ is the variance of the post-detection noise; and

$SNR_{mmse,\ell}(k)$ is the post-detection SNR for subband k of transmit antenna ℓ.

The post-detection SNRs for later stages improve because the norm of $\underline{\mathbf{w}}_{mmse,\ell}(k)$ in equation (22) decreases with each stage.

[0069] The SIC technique is described in further detail in commonly assigned U.S. Patent Application Serial No. 09/993,087, entitled "Multiple-Access Multiple-Input Multiple-Output (MIMO) Communication System," filed November 6, 2001.

[0070] For the second transmission scheme, the receiver can also recover the N_T data symbol streams using an iterative detection and decoding (IDD) scheme. For the IDD scheme, whenever a block of received symbols for a data packet is obtained, the receiver iteratively performs detection and decoding multiple (N_{dec}) times on the received symbols in the block to obtain a decoded packet. A detector performs detection on the received symbol block and provides a detected symbol block. A decoder performs decoding on the detected symbol block and provides decoder *a priori* information, which is used by the detector in a subsequent iteration. The decoded packet is generated based on decoder output for the last iteration.

[0071] It can be shown that the first transmission scheme and the second transmission scheme with either the SIC with MMSE receiver or the IDD receiver are optimal and can achieve capacity or near capacity for the MIMO-OFDM system. The second transmission scheme with a maximum likelihood detector for the received symbols can also provide optimal or near optimal performance. Other capacity achieving transmission schemes may also be used for the MIMO-OFDM system. One such capacity achieving transmission scheme is an autocoding scheme described by T.L. Marzetta *et al.* in a paper entitled "Structured Unitary Space-Time Autocoding Constellations," IEEE Transaction on Information Theory, Vol. 48, No. 4, April 2002.

[0072] **FIG. 5** shows a block diagram of transmitter 110. Within TX data processor 120, an encoder 520 receives and encodes a data stream $\{d\}$ in accordance with coding scheme C for the selected rate R and provides code bits. The encoding increases the reliability of the data transmission. The coding scheme may include a convolutional code, a Turbo code, a block code, a CRC code, or a combination thereof. A channel interleaver 522 interleaves (i.e., reorders) the code bits from encoder 520 based on an interleaving scheme. The interleaving provides time and/or frequency diversity for the code bits. A symbol mapping unit 524 modulates (i.e., symbol maps) the interleaved data from channel interleaver 522 in accordance with modulation scheme M for the selected rate R and provides data symbols. The modulation may be achieved by (1) grouping sets of B interleaved bits to form B-bit binary values, where $B \geq 1$, and (2) mapping each B-bit binary value to a specific signal point in a signal constellation for

the modulation scheme. Symbol mapping unit 524 provides a stream of data symbols $\{s\}$.

[0073] Transmitter 110 encodes and modulates each data packet separately based on the rate R selected for the packet to obtain a corresponding data symbol block. Transmitter 110 may transmit one data symbol block at a time on all subbands of all spatial channels available for data transmission. Each data symbol block may be transmitted in one or multiple OFDM symbol periods. Transmitter 110 may also transmit multiple data symbol blocks simultaneously on the available subbands and spatial channels. If one rate is selected for each time interval, as described above, then all data symbol block(s) transmitted in the same time interval use the same selected rate.

[0074] For the embodiment shown in FIG. 5, TX spatial processor 130 implements the second transmission scheme described above. Within TX spatial processor 130, a multiplexer/demultiplexer (Mux/Demux) 530 receives and demultiplexes the data symbol stream $\{s\}$ into N_T streams for the N_T transmit antennas. Mux/demux 530 also multiplexes in pilot symbols (e.g., in a time division multiplex (TDM) manner) and provides N_T transmit symbol streams, $\{x_1\}$ through $\{x_{N_T}\}$, for the N_T transmit antennas. Each transmit symbol may be a data symbol, a pilot symbol, or a signal value of zero for a subband not used for data or pilot transmission.

[0075] Transmitter unit 132 includes N_T OFDM modulators 532a through 532t and N_T TX RF units 534a through 534t for the N_T transmit antennas. Each OFDM modulator 532 performs OFDM modulation on a respective transmit symbol stream by (1) grouping and transforming each set of N_F transmit symbols for the N_F subbands to the time domain using an N_F-point IFFT to obtain a corresponding transformed symbol that contains N_F chips and (2) repeating a portion (or N_{cp} chips) of each transformed symbol to obtain a corresponding OFDM symbol that contains $N_F + N_{cp}$ chips. The repeated portion is referred to as a cyclic prefix, which ensures that the OFDM symbol retains its orthogonal properties in the presence of delay spread in a multipath channel. Each OFDM modulator 532 provides a stream of OFDM symbols, which is further conditioned (e.g., converted to analog, frequency upconverted, filtered, and amplified) by an associated TX RF unit 534 to generate a modulated signal. The N_T modulated signals from TX RF units 534a through 534t are transmitted from N_T antennas 540a through 540t, respectively.

[0076] FIG. 6 shows a block diagram of receiver 150. N_R receive antennas 652a through 652r receive the modulated signals transmitted by transmitter 110 and provide N_R received signals to receiver unit 154. Receiver unit 154 includes N_R RX RF units 654a through 654r and N_R OFDM demodulators 656a through 656r for the N_R receive antennas. Each RX RF unit 654 conditions and digitizes a respective received signal and provides a stream of samples. Each OFDM demodulator 656 performs OFDM demodulation on a respective sample stream by (1) removing the cyclic prefix in each received OFDM symbol to obtain a received transformed symbol and (2) transforming each received transformed symbol to the frequency domain with an N_F-point FFT to obtain N_F received symbols for the N_F subbands. Each OFDM demodulator 656 provides received data symbols to RX spatial processor 160 and received pilot symbols to channel estimator 172.

[0077] FIG. 6 also shows an RX spatial processor 160a and an RX data processor 170a, which are one embodiment of RX spatial processor 160 and RX data processor 170, respectively, at receiver 150. Within RX spatial processor 160a, a detector 660 performs spatial processing/detection on the N_R received symbol streams to obtain N_T detected symbol streams. Each detected symbol is an estimate of a data symbol transmitted by the transmitter. Detector 660 may implement an MMSE, MRC, or zero-forcing detector. The detection is performed for each subband based on a matched filter matrix (or detector response) $\underline{\mathbf{W}}(k)$ for that subband, which is derived based on an estimate of the channel response matrix $\underline{\mathbf{H}}(k)$ for the subband. For example, the matched filter matrix for the MMSE detector may be derived as: $\underline{\mathbf{W}}_{mmse}(k) = (\underline{\mathbf{H}}(k) \cdot \underline{\mathbf{H}}^H(k) + N_0 \cdot \underline{\mathbf{I}})^{-1} \cdot \underline{\mathbf{H}}(k)$. A multiplexer 662 multiplexes the detected symbols and provides a detected symbol stream $\{\hat{s}\}$ to RX data processor 170a.

[0078] Within RX data processor 170a, a symbol demapping unit 670 demodulates the detected symbols in accordance with the modulation scheme M for the selected rate R and provides demodulated data. A channel deinterleaver 672 deinterleaves the demodulated data in a manner complementary to the interleaving performed at the transmitter and provides deinterleaved data. A decoder 674 decodes the deinterleaved data in a manner complementary to the encoding performed at the transmitter and provides a decoded data stream $\{\hat{d}\}$. For example, decoder 674 may implement a Turbo decoder or a Viterbi decoder if Turbo or convolutional coding, respectively, is

performed at the transmitter. Decoder 674 also provides the status of each decoded packet, which indicates whether the packet is decoded correctly or in error.

[0079] **FIG. 7** shows an RX spatial processor 160b and an RX data processor 170b, which implement the IDD scheme and are another embodiment of RX spatial processor 160 and RX data processor 170, respectively, at receiver 150. A detector 760 and a decoder 780 perform iterative detection and decoding on the received symbols for each data packet to obtain a decoded packet. The IDD scheme exploits the error correction capabilities of the channel code to provide improved performance. This is achieved by iteratively passing *a priori* information between detector 760 and decoder 780 for N_{dec} iterations, where $N_{dec} > 1$. The *a priori* information indicates the likelihood of each transmitted data bit being zero or one.

[0080] Within RX spatial processor 160b, a buffer 758 receives and stores N_R received symbol sequences from the N_R receive antennas for each data packet. The iterative detection and decoding process is performed on each block of received symbols for a data packet. Detector 760 performs spatial processing on the N_R received symbol sequences for each block and provides N_T detected symbol sequences for the block. Detector 760 may implement an MMSE, MRC, or zero-forcing detector. A multiplexer 762 multiplexes the detected symbols in the N_T sequences and provides a detected symbol block.

[0081] Within RX data processor 170b, a log-likelihood ratio (LLR) computation unit 770 receives the detected symbols from RX spatial processor 160b and computes the LLRs for the B code bits of each detected symbol. These LLRs represent *a priori* information provided by detector 760 to decoder 780. A channel deinterleaver 772 deinterleaves each block of LLRs from LLR computation unit 770 and provides deinterleaved LLRs $\{x^n\}$ for the block. Decoder 780 decodes the deinterleaved LLRs and provides decoder LLRs $\{x^{n+1}\}$, which represent *a priori* information provided by decoder 780 to detector 760. The decoder LLRs are interleaved by a channel interleaver 782 and provided to detector 760.

[0082] The detection and decoding process is then repeated for another iteration. Detector 760 derives new detected symbols based on the received symbols and the decoder LLRs. The new detected symbols are again decoded by decoder 780. The detection and decoding process is iterated N_{dec} times. During the iterative detection and decoding process, the reliability of the detected symbols improves with each

detection/decoding iteration. After all N_{dec} detection/decoding iterations have been completed, decoder 780 computes the final data bit LLRs and slices these LLRs to obtain the decoded packet.

[0083] The IDD scheme is described in further detail in commonly assigned U.S. Patent Application Serial No. 60/506,466, entitled "Hierarchical Coding With Multiple Antennas in a Wireless Communication System," filed September 25, 2003.

[0084] The rate selection and data transmission techniques described herein may be implemented by various means. For example, these techniques may be implemented in hardware, software, or a combination thereof. For a hardware implementation, the processing units used to perform rate selection and data transmission may be implemented within one or more application specific integrated circuits (ASICs), digital signal processors (DSPs), digital signal processing devices (DSPDs), programmable logic devices (PLDs), field programmable gate arrays (FPGAs), processors, controllers, micro-controllers, microprocessors, other electronic units designed to perform the functions described herein, or a combination thereof.

[0085] For a software implementation, the rate selection and data transmission techniques may be implemented with modules (e.g., procedures, functions, and so on) that perform the functions described herein. The software codes may be stored in a memory unit (e.g., memory unit 182 or 142 in FIG. 1) and executed by a processor (e.g., controller 180 or 140). The memory unit may be implemented within the processor or external to the processor, in which case it can be communicatively coupled to the processor via various means as is known in the art.

[0086] The previous description of the disclosed embodiments is provided to enable any person skilled in the art to make or use the present invention. Various modifications to these embodiments will be readily apparent to those skilled in the art, and the generic principles defined herein may be applied to other embodiments without departing from the spirit or scope of the invention. Thus, the present invention is not intended to be limited to the embodiments shown herein but is to be accorded the widest scope consistent with the principles and novel features disclosed herein.

[0087] **WHAT IS CLAIMED IS:**

CLAIMS

1. A method of selecting a rate for data transmission in a multi-carrier multiple-input multiple-output (MIMO) communication system, comprising:

determining an average constrained spectral efficiency for a plurality of subbands of a plurality of spatial channels used for data transmission, the plurality of spatial channels being formed by a MIMO channel in the system;

determining an equivalent signal-to-noise-and-interference ratio (SNR) needed by an equivalent system with an additive white Gaussian noise (AWGN) channel to support the average constrained spectral efficiency; and

selecting the rate for data transmission in the multi-carrier MIMO system based on the equivalent SNR.

2. The method of claim 1, wherein the average constrained spectral efficiency, the equivalent SNR, and the rate are all determined based on a specific modulation scheme.

3. The method of claim 1, wherein the plurality of subbands are obtained with orthogonal frequency division multiplexing (OFDM).

4. The method of claim 1, wherein the plurality of spatial channels correspond to a plurality of single-input single-output (SISO) channels that make up the MIMO channel.

5. The method of claim 1, further comprising:

determining a post-detection SNR for each subband of each spatial channel used for data transmission; and

determining a constrained spectral efficiency for each subband of each spatial channel based on the post-detection SNR for the subband of the spatial channel, and

wherein the average constrained spectral efficiency is determined based on constrained spectral efficiencies for the plurality of subbands of the plurality of spatial channels.

6. The method of claim 5, wherein the post-detection SNR for each subband of each spatial channel is determined based on a transmission scheme capable of achieving capacity of the MIMO channel.

7. The method of claim 5, wherein the post-detection SNR for each subband of each spatial channel is determined based on successive interference cancellation (SIC) processing with a minimum mean square error (MMSE) detector at a receiver.

8. The method of claim 5, wherein the constrained spectral efficiency for each subband of each spatial channel is further determined based on a constrained spectral efficiency function having an SNR and a modulation scheme as inputs and providing a constrained spectral efficiency as output.

9. The method of claim 1, further comprising:

determining a constrained spectral efficiency for each subband of the MIMO channel based on a constrained spectral efficiency function having a MIMO channel response and a modulation scheme as inputs and providing a constrained spectral efficiency as output, and

wherein the average constrained spectral efficiency for the plurality of subbands of the plurality of spatial channels used for data transmission is determined based on constrained spectral efficiencies for the plurality of subbands of the MIMO channel.

10. The method of claim 1, wherein the equivalent SNR is determined based on an inverse constrained spectral efficiency function having a spectral efficiency and a modulation scheme as inputs and providing an SNR as output.

11. The method of claim 1, wherein the rate for data transmission is selected based on a set of rates supported by the multi-carrier MIMO system and required SNRs for the supported rates.

12. The method of claim 11, wherein the selected rate is a highest rate among the supported rates having a required SNR less than or equal to the equivalent SNR.

13. The method of claim 11, wherein the required SNRs for the supported rates include losses observed by the multi-carrier MIMO system.

14. The method of claim 1, further comprising:

determining a back-off factor to account for error in rate prediction and system losses; and

reducing the rate for data transmission based on the back-off factor.

15. The method of claim 1, further comprising:

receiving a data transmission at the selected rate, wherein the received data transmission includes at least one block of data symbols for at least one data packet, and wherein the data symbols in each block are transmitted simultaneously on the plurality of subbands of the plurality of spatial channels used for data transmission.

16. The method of claim 1, further comprising:

receiving a data transmission at the selected rate; and

performing iterative detection and decoding (IDD) to recover data in the received data transmission.

17. An apparatus in a multi-carrier multiple-input multiple-output (MIMO) communication system, comprising:

a channel estimator operative to obtain channel estimates for a MIMO channel in the system; and

a controller operative to

determine an average constrained spectral efficiency for a plurality of subbands of a plurality of spatial channels used for data transmission based on the channel estimates, wherein the plurality of spatial channels are formed by the MIMO channel,

determine an equivalent signal-to-noise-and-interference ratio (SNR) needed by an equivalent system with an additive white Gaussian noise (AWGN) channel to support the average constrained spectral efficiency, and

select a rate for data transmission in the multi-carrier MIMO system based on the equivalent SNR.

18. The apparatus of claim 17, wherein the controller is further operative to

determine a post-detection SNR for each subband of each spatial channel used for data transmission based on the channel estimates, and

determine a constrained spectral efficiency for each subband of each spatial channel based on the post-detection SNR for the subband of the spatial channel, and wherein the average constrained spectral efficiency is determined based on constrained spectral efficiencies for the plurality of subbands of the plurality of spatial channels.

19. The apparatus of claim 18, wherein the post-detection SNR for each subband of each spatial channel is further determined based on a transmission scheme capable of achieving capacity of the MIMO channel.

20. The apparatus of claim 17, wherein a set of rates is supported by the multi-carrier MIMO system and each supported rate is associated with a respective required SNR, and wherein the controller is further operative to select a highest rate among the supported rates having a required SNR less than or equal to the equivalent SNR.

21. The apparatus of claim 17, wherein the controller is further operative to determine a back-off factor to account for error in rate prediction and system losses and to reduce the rate for data transmission based on the back-off factor.

22. The apparatus of claim 17, further comprising:

a receive spatial processor operative to perform detection on received symbols for a data transmission at the selected rate and provide detected symbols; and

a receive data processor operative to process the detected symbols to obtain decoded data.

23. The apparatus of claim 22, wherein the receive spatial processor and the receive data processor are operative to perform iterative detection and decoding (IDD) to obtain the decoded data from the received symbols.

24. An apparatus in a multi-carrier multiple-input multiple-output (MIMO) communication system, comprising:

means for determining an average constrained spectral efficiency for a plurality of subbands of a plurality of spatial channels used for data transmission, the plurality of spatial channels being formed by a MIMO channel in the system;

means for determining an equivalent signal-to-noise-and-interference ratio (SNR) needed by an equivalent system with an additive white Gaussian noise (AWGN) channel to support the average constrained spectral efficiency; and

means for selecting a rate for data transmission in the multi-carrier MIMO system based on the equivalent SNR.

25. The apparatus of claim 24, further comprising:

means for determining a post-detection SNR for each subband of each spatial channel used for data transmission; and

means for determining a constrained spectral efficiency for each subband of each spatial channel based on the post-detection SNR for the subband of the spatial channel, and wherein the average constrained spectral efficiency is determined based on constrained spectral efficiencies for the plurality of subbands of the plurality of spatial channels.

26. The apparatus of claim 24, further comprising:

means for determining a back-off factor to account for error in rate prediction and system losses; and

means for reducing the rate for data transmission based on the back-off factor.

27. The apparatus of claim 24, further comprising:

means for receiving a data transmission at the selected rate; and

means for performing iterative detection and decoding (IDD) to recover data in the received data transmission.

28. A processor readable media for storing instructions operable in an apparatus to:

determine an average constrained spectral efficiency for a plurality of subbands of a plurality of spatial channels used for data transmission in a multi-carrier multiple-

input multiple-output (MIMO) communication system, the plurality of spatial channels being formed by a MIMO channel in the system;

determine an equivalent signal-to-noise-and-interference ratio (SNR) needed by an equivalent system with an additive white Gaussian noise (AWGN) channel to support the average constrained spectral efficiency; and

select a rate for data transmission in the multi-carrier MIMO system based on the equivalent SNR.

29. The processor readable media of claim 28 and further for storing instructions operable to

determine a post-detection SNR for each subband of each spatial channel used for data transmission; and

determine a constrained spectral efficiency for each subband of each spatial channel based on the post-detection SNR for the subband of the spatial channel, and wherein the average constrained spectral efficiency is determined based on constrained spectral efficiencies for the plurality of subbands of the plurality of spatial channels.

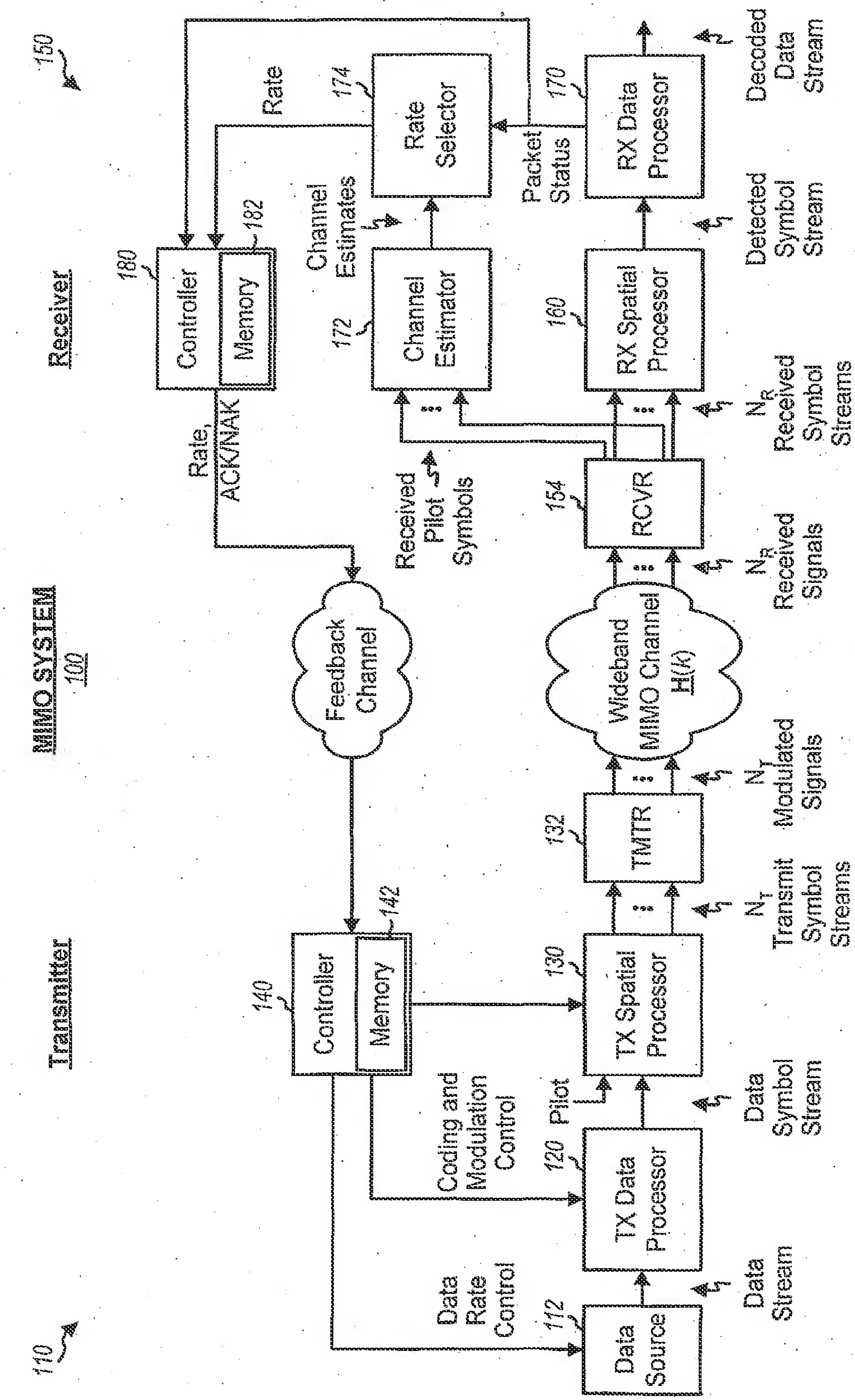

MIMO SYSTEM
100
Transmitter
Receiver
110
150
Data Source
112
Data Rate Control
Data Stream
TX Data Processor
120
Coding and Modulation Control
Pilot
Data Symbol Stream
TX Spatial Processor
130
N_T Transmit Symbol Streams
TMTR
132
N_T Modulated Signals
Controller
140
Memory
142
Feedback Channel
Wideband MIMO Channel $\underline{H}(k)$
N_R Received Signals
RCVR
154
Received Pilot Symbols
N_R Received Symbol Streams
Rate, ACK/NAK
Controller
180
Memory
182
Channel Estimator
172
Channel Estimates
RX Spatial Processor
160
Detected Symbol Stream
Rate Selector
174
Rate
Packet Status
RX Data Processor
170
Decoded Data Stream

FIG. 1

2/8

R = rate
M = modulation scheme
C = coding scheme
P_e = probability of error
ℓ = index for spatial channels
k = index for subbands
S = spectral efficiency

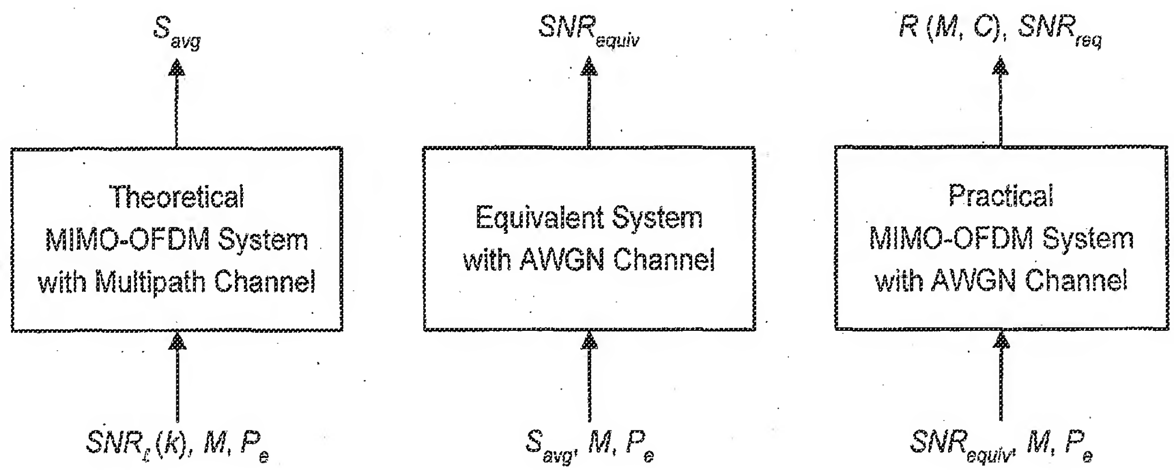


FIG. 2

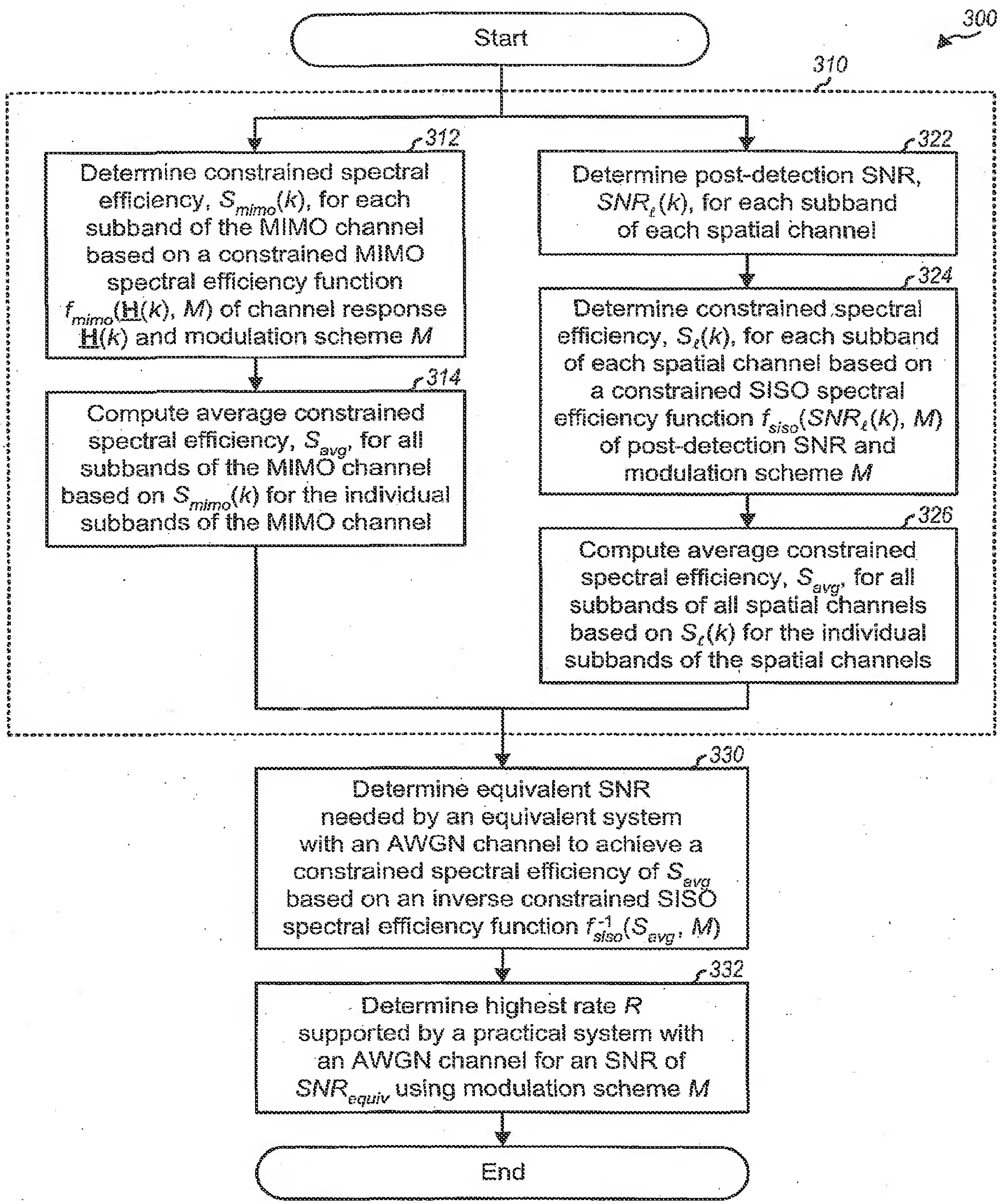

300
Start
310
312
Determine constrained spectral efficiency, $S_{mimo}(k)$, for each subband of the MIMO channel based on a constrained MIMO spectral efficiency function $f_{mimo}(\underline{\mathbf{H}}(k), M)$ of channel response $\underline{\mathbf{H}}(k)$ and modulation scheme M
314
Compute average constrained spectral efficiency, S_{avg}, for all subbands of the MIMO channel based on $S_{mimo}(k)$ for the individual subbands of the MIMO channel
322
Determine post-detection SNR, $SNR_{\ell}(k)$, for each subband of each spatial channel
324
Determine constrained spectral efficiency, $S_{\ell}(k)$, for each subband of each spatial channel based on a constrained SISO spectral efficiency function $f_{siso}(SNR_{\ell}(k), M)$ of post-detection SNR and modulation scheme M
326
Compute average constrained spectral efficiency, S_{avg}, for all subbands of all spatial channels based on $S_{\ell}(k)$ for the individual subbands of the spatial channels
330
Determine equivalent SNR needed by an equivalent system with an AWGN channel to achieve a constrained spectral efficiency of S_{avg} based on an inverse constrained SISO spectral efficiency function $f_{siso}^{-1}(S_{avg}, M)$
332
Determine highest rate R supported by a practical system with an AWGN channel for an SNR of SNR_{equiv} using modulation scheme M
End

FIG. 3

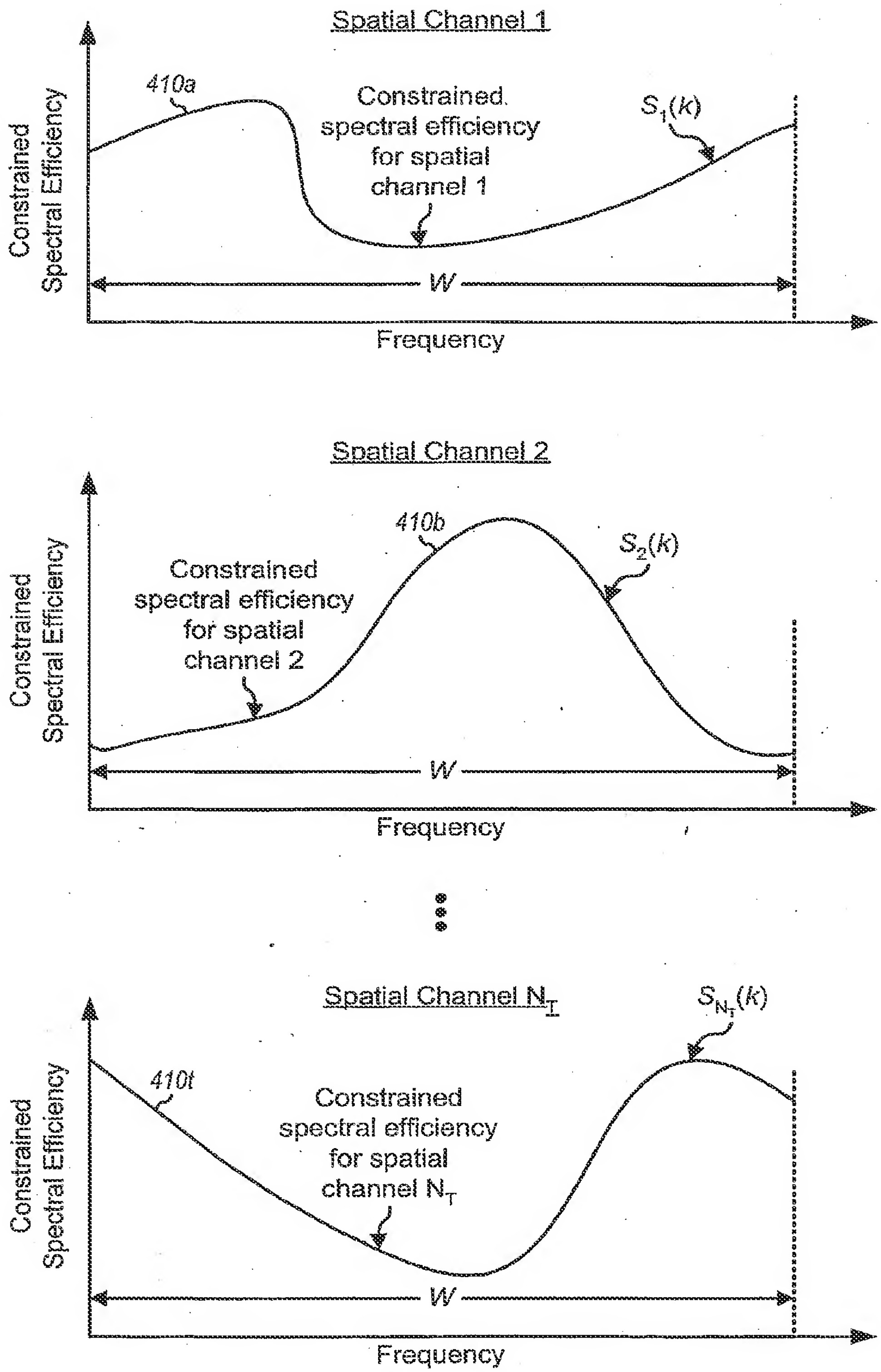

Spatial Channel 1
Constrained Spectral Efficiency
410a
Constrained spectral efficiency for spatial channel 1
S1(k)
W
Frequency
Spatial Channel 2
Constrained Spectral Efficiency
410b
Constrained spectral efficiency for spatial channel 2
S2(k)
W
Frequency
Spatial Channel NT
SNT(k)
Constrained Spectral Efficiency
410t
Constrained spectral efficiency for spatial channel NT
W
Frequency

FIG. 4A

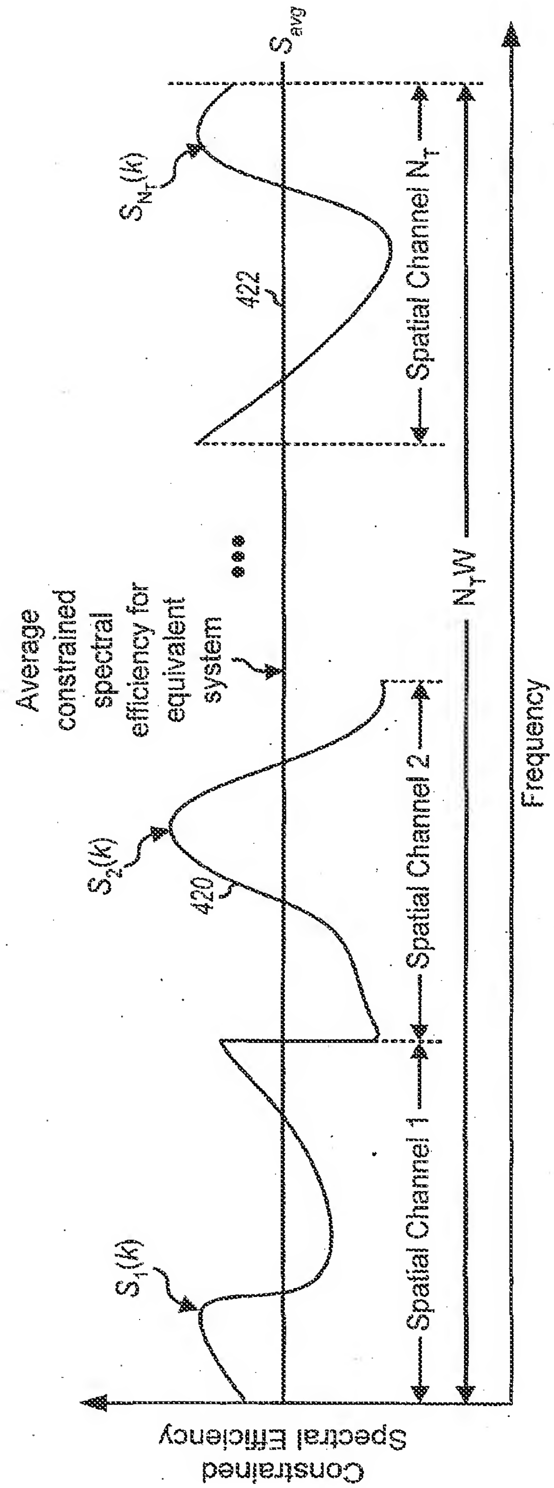

Constrained Spectral Efficiency
S_{avg}
$S_1(k)$
$S_2(k)$
$S_{N_T}(k)$
420
422
Average constrained spectral efficiency for equivalent system
Spatial Channel 1
Spatial Channel 2
Spatial Channel N_T
$N_T W$
Frequency

FIG. 4B

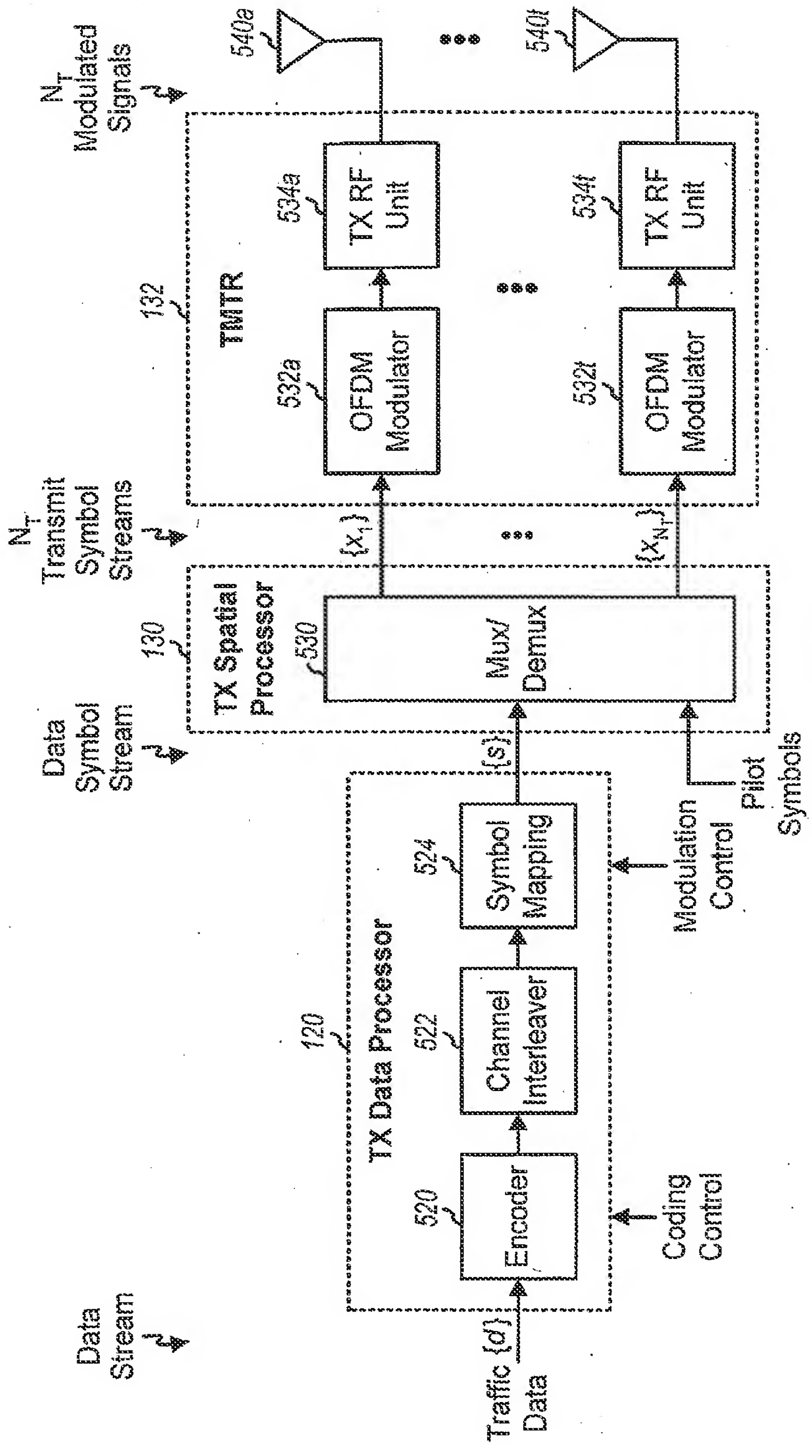
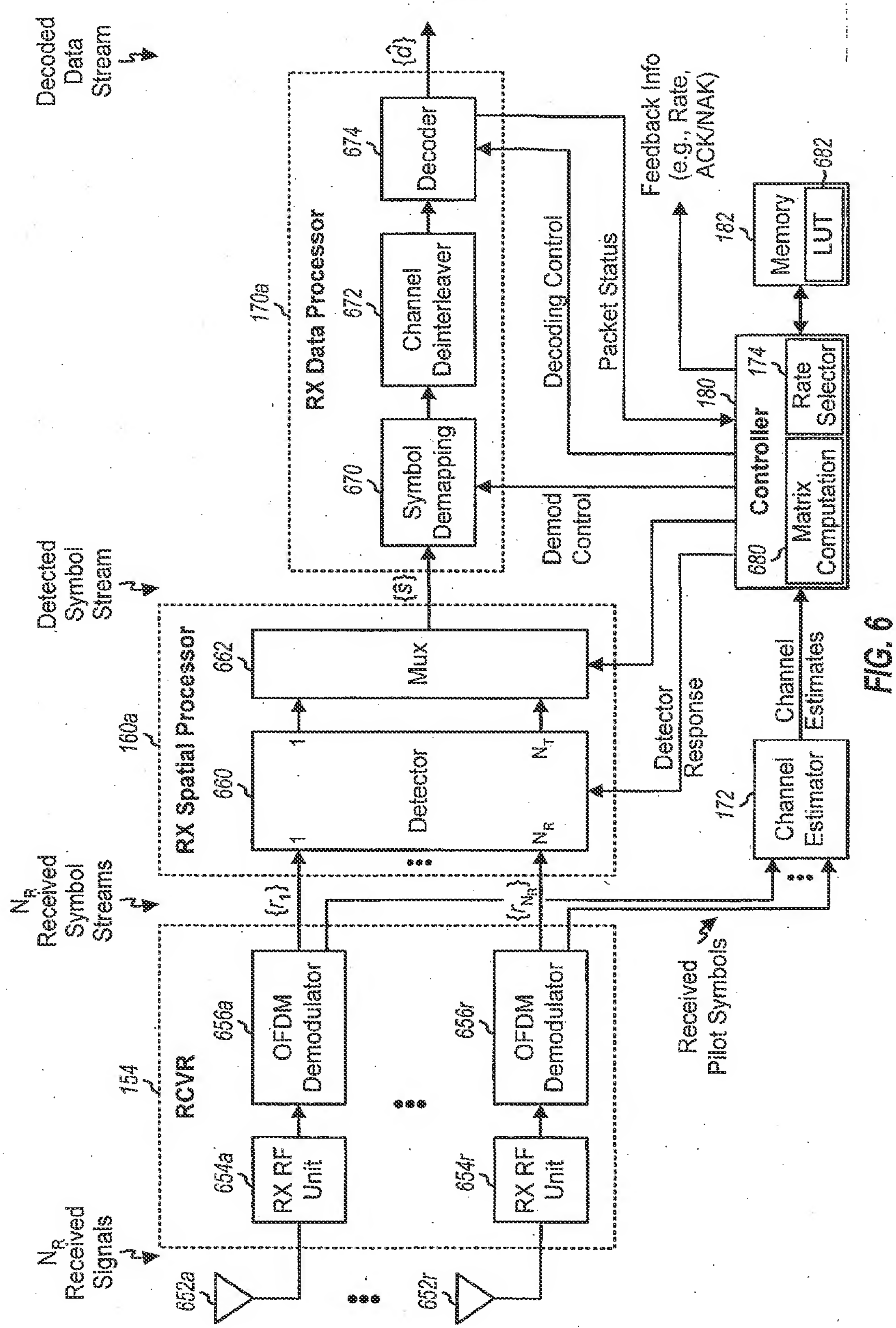
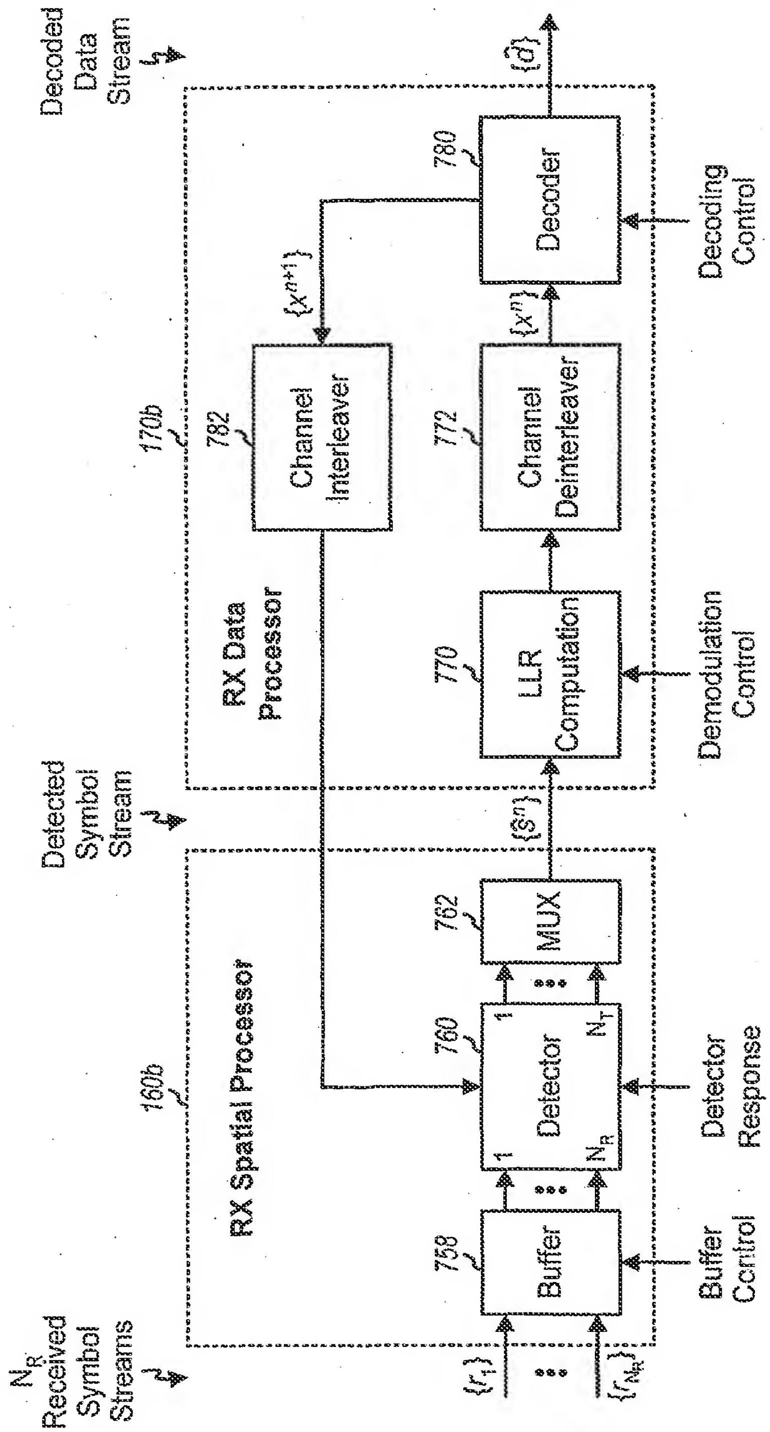

Data Stream
Data Symbol Stream
N_T Transmit Symbol Streams
N_T Modulated Signals
Traffic Data
{d}
120
TX Data Processor
520
Encoder
522
Channel Interleaver
524
Symbol Mapping
Coding Control
Modulation Control
{s}
Pilot Symbols
130
TX Spatial Processor
530
Mux/ Demux
$\{x_1\}$
$\{x_{N_T}\}$
132
TMTR
532a
OFDM Modulator
534a
TX RF Unit
532t
OFDM Modulator
534t
TX RF Unit
540a
540t

FIG. 5



N_R Received Signals
652a
652r
154
RCVR
654a
RX RF Unit
654r
RX RF Unit
656a
OFDM Demodulator
656r
OFDM Demodulator
N_R Received Symbol Streams
$\{r_1\}$
$\{r_{N_R}\}$
Received Pilot Symbols
160a
RX Spatial Processor
660
Detector
1
N_R
1
N_T
662
Mux
Detector Response
Detected Symbol Stream
$\{\hat{s}\}$
170a
RX Data Processor
670
Symbol Demapping
672
Channel Deinterleaver
674
Decoder
$\{\hat{d}\}$
Decoded Data Stream
Demod Control
Decoding Control
Packet Status
172
Channel Estimator
Channel Estimates
180
Controller
680
Matrix Computation
174
Rate Selector
182
Memory
LUT
682
Feedback Info (e.g., Rate, ACK/NAK)

FIG. 6



N_R Received Symbol Streams
$\{r_1\}$
$\{r_{N_R}\}$
160b
RX Spatial Processor
758
Buffer
Buffer Control
760
Detector
1
N_R
1
N_T
Detector Response
762
MUX
Detected Symbol Stream
$\{\hat{s}^n\}$
170b
RX Data Processor
770
LLR Computation
Demodulation Control
772
Channel Deinterleaver
$\{x^n\}$
780
Decoder
Decoding Control
$\{\hat{d}\}$
Decoded Data Stream
782
Channel Interleaver
$\{x^{n+1}\}$

FIG. 7

INTERNATIONAL SEARCH REPORT

International Application No
PCT/US2004/033680

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04L27/26 H04L1/06

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	WO 03/047198 A (QUALCOMM INCORPORATED) 5 June 2003 (2003-06-05) abstract page 3, paragraph 1018 page 7, paragraph 1035 - paragraph 1038 page 8, paragraph 1041 - page 12, paragraph 1058 page 15, paragraph 1073	1-29

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[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

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Date of the actual completion of the international search	Date of mailing of the international search report
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C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	KA-WAI NG ET AL: "A simplified bit allocation for V-BLAST based OFDM MIMO systems in frequency selective fading channels" ICC 2002. 2002 IEEE INTERNATIONAL CONFERENCE ON COMMUNICATIONS. CONFERENCE PROCEEDINGS. NEW YORK, NY, APRIL 28 - MAY 2, 2002, IEEE INTERNATIONAL CONFERENCE ON COMMUNICATIONS, NEW YORK, NY : IEEE, US, vol. VOL. 1 OF 5, 28 April 2002 (2002-04-28), pages 411-415, XP010589527 ISBN: 0-7803-7400-2 abstract II. System overview IV. Proposed allocation algorithm	1-29
A	WO 03/075479 A (QUALCOMM INCORPORATED) 12 September 2003 (2003-09-12) page 1, paragraph 1001 page 2, paragraph 1007 - page 3, paragraph 1008 page 6, paragraph 1024 - page 7, paragraph 1031 page 12, line 1044 - page 14, line 1052 page 15, paragraph 1056 - page 16, paragraph 1059 page 19, paragraph 1069 page 21, paragraph 1081 - page 22, paragraph 1082	1-29
A	US 6 141 317 A (MARCHOK DANIEL J 'US! ET AL) 31 October 2000 (2000-10-31) column 2, line 5 - line 27 column 21, line 45 - column 23, line 30	1-29

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/US2004/033680

Patent document cited in search report		Publication date	Patent family member(s)			Publication date
WO 03047198	A	05-06-2003	US	2003095506	A1	22-05-2003
			US	2003095508	A1	22-05-2003
			AU	2002346470	A1	10-06-2003
			AU	2002365351	A1	10-06-2003
			EP	1454466	A2	08-09-2004
			EP	1454467	A2	08-09-2004
			WO	03047197	A2	05-06-2003
			WO	03047198	A2	05-06-2003
WO 03075479	A	12-09-2003	US	2003165189	A1	04-09-2003
			AU	2003216479	A1	16-09-2003
			CA	2476061	A1	12-09-2003
			EP	1481489	A1	01-12-2004
			WO	03075479	A1	12-09-2003
			US	2004037235	A1	26-02-2004
US 6141317	A	31-10-2000	US	5790514	A	04-08-1998
			US	6473394	B1	29-10-2002
			US	6108349	A	22-08-2000
			US	6804192	B1	12-10-2004
			US	6690680	B1	10-02-2004
			US	6771590	B1	03-08-2004
			US	6118758	A	12-09-2000
			US	2004184484	A1	23-09-2004
			US	6285654	B1	04-09-2001
			US	2004246890	A1	09-12-2004
			US	6122246	A	19-09-2000
			US	5995483	A	30-11-1999
			US	2002034160	A1	21-03-2002

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A1

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(30) Priority Data:
118,473 8 September 1993 (08.09.93) US

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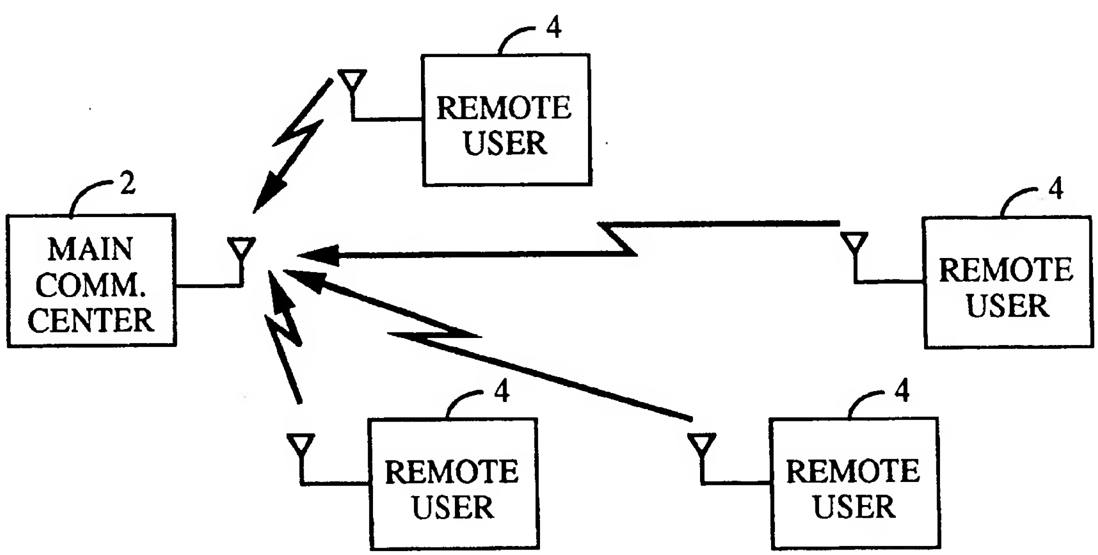
(74) Agent: MILLER, Russell, B.; Qualcomm Incorporated, 6455 Lusk Boulevard, San Diego, CA 92121 (US).

(81) Designated States: AM, AT, AU, BB, BG, BR, BY, CA, CH, CN, CZ, DE, DK, EE, ES, FI, GB, GE, HU, JP, KE, KG, KP, KR, KZ, LK, LR, LT, LU, LV, MD, MG, MN, MW, NL, NO, NZ, PL, PT, RO, RU, SD, SE, SI, SK, TJ, TT, UA, UZ, VN, European patent (AT, BE, CH, DE, DK, ES, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, ML, MR, NE, SN, TD, TG), ARIPO patent (KE, MW, SD).

Published
With international search report.

(54) Title: METHOD AND APPARATUS FOR DETERMINING THE TRANSMISSION DATA RATE IN A MULTI-USER COMMUNICATION SYSTEM



(57) Abstract

A method and apparatus for controlling the data rates for communications to and from a base station (2) and a plurality of remote users (4). The usage of the communications resource whether the forward link resource, from base station (2) to remote users (4), or reverse link resource, from remote users (4) to base station (2), is measured. The measured usage value is compared against at least one predetermined threshold value and the data rates of communications or a subset of communications on said communications resource is modified in accordance with said comparisons.

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METHOD AND APPARATUS FOR DETERMINING THE TRANSMISSION DATA RATE IN A MULTI-USER COMMUNICATION SYSTEM

BACKGROUND OF THE INVENTION

I. Field of the Invention

The present invention relates to communications systems. More particularly, the present invention relates to a novel and improved method and apparatus for maximizing total average service quality to users in a multi-user communication system by controlling the data transmission rates to and from users of the multi-user communication system.

II. Description of the Related Art

The term "multiple access" refers to the sharing of a fixed communications resource by a plurality of users. A typical example of such a fixed communications resource is bandwidth. There are three basic ways to increase the throughput or data rate of an individual user accessing a communications resource. The first way is to increase the transmitters radiated power or alternatively to reduce system losses so that the received signal to noise ratio (SNR) is increased. The second way is to increase the allocation of bandwidth to the user. The third approach is to make allocation of the communications resource more efficient.

Some of the more common methods of providing multiple access to a communications resource involve both analog and digital communication modulation schemes. Such schemes include frequency division, time division and spread spectrum techniques. In frequency division multiple access (FDMA) techniques, each user is allocated one or more specific sub-bands of frequency. In time division multiple access (TDMA) techniques, periodically recurring time slots are identified, and for each segment of time each user is allocated one or more time slots. In some TDMA systems, users are provided a fixed assignment in time, and in other systems users may access the resource at random times. In spread spectrum communications, users share a common frequency band. Using frequency hopping (FH) modulation, the signal is modulated upon a carrier which changes in frequency according to a predetermined plan. In direct sequence (DS) modulation, the user signal is modulated with a pseudorandom code. In one type of code division multiple access (CDMA) technique which uses

direct sequence spread spectrum modulation, a set of orthogonal or nearly orthogonal spread spectrum codes (each using full channel bandwidth) are identified, and each user is allocated one or more specified codes.

In all multiple access schemes, a plurality of users shares a communications resource without creating unmanageable interference to each other in the detection process. The allowable limit of such interference is defined to be the maximum amount of interference such that the resulting transmission quality is still above a predetermined acceptable level. In digital transmission schemes, the quality is often measured by the bit error rate (BER) or frame error rate (FER). In digital speech communications systems, the overall speech quality is limited by data rate allowed for each user, and by the BER or FER.

Systems have been developed to minimize the data rate required for a speech signal while still providing an acceptable level of speech quality. If speech is transmitted by simply sampling and digitizing the analog speech signal, a data rate on the order of 64 kilobits per second (Kbps) is required to achieve a speech quality equivalent to that of a conventional analog telephone. However, through the use of speech analysis, followed by the appropriate coding, transmission, and resynthesis at the receiver, a significant reduction in the data rate can be achieved with a minimal decrease in quality.

Devices which employ techniques to compress speech by extracting parameters that relate to a model of human speech generation are typically called vocoders. Such devices are composed of an encoder, which analyzes the incoming speech to extract the relevant parameters, and a decoder, which resynthesizes the speech using the parameters which are received from the encoder over the transmission channel. As the speech changes, new model parameters are determined and transmitted over the communications channel. The speech is typically segmented into blocks of time, or analysis frames, during which the parameters are calculated. The parameters are then updated for each new frame.

A more preferred technique to accomplish data compression, so as to result in a reduction of information that needs to be sent, is to perform variable rate vocoding. An example of variable rate vocoding is detailed in U.S. Patent Application Serial No. 08/004,484 entitled "Variable Rate Vocoder," assigned to the assignee of the present invention and incorporated herein by reference. Since speech inherently contains periods of silence, i.e. pauses, the amount of data required to represent these periods can be reduced. Variable rate vocoding most effectively exploits this fact by

reducing the data rate for these periods of silence. A reduction in the data rate, as opposed to a complete halt in data transmission, for periods of silence overcomes the problems associated with voice activity gating while facilitating a reduction in transmitted information, thus reducing the overall interference in a multiple access communication system.

It is the objective of the present invention to modify the variability of the transmission rate of variable rate vocoders, and any other variable rate data source, in order to maximize utilization of the communications resource.

SUMMARY OF THE INVENTION

The present invention is a novel and improved method and apparatus for maximizing total average service quality to users in a multi-user communication system by controlling the data transmission rates to and from users of the multi-user communication system.

In the present invention, usage of the available communication resource is monitored. When the usage of the available communication resource becomes too great for a given communications link, and thus the quality falls below a predetermined limit, the data rate to or from the users is limited to free up a portion of the available communication resource. When the usage of the communications resource becomes small, the data rate to or from the users is allowed to increase above the previous limit.

For example, if the communications link from remote users to a main communications center, hereafter known as the reverse link, becomes overloaded, the main communications center transmits a signaling message requesting that the users, or selected ones of the users, decrease their average transmission data rate. At the remote user end, the signaling message is received and the transmission rate for the remote user is lowered in accordance with the signaling message.

The remote user, in the example, may be transmitting speech data or other digital data. If the user is transmitting speech data, then his transmission data rate may be adjusted using a variable rate vocoder as is described in above mentioned Application Serial No. 08/004,484. The present invention is equally applicable to any variable rate vocoding strategy when the remote user is transmitting speech data. If the user is transmitting digital data that is not speech data, the system can optionally instruct the remote user to modify the transmitted data rate for the specific digital data source.

On the communication link between the main communication center and the remote users, hereafter known as the forward link, the main communication center monitors the fraction of its total resource capacity that is being used for communicating to the remote users. If the fraction of the communications resource being used is too large, the main communication center will decrease the permitted average transmission data rate to each user or a subset of users. If the fraction of the communications resource being used is too small, the main communication center will permit the average data rate for each user to increase. As in the reverse link, the control of the data rate may be selective in nature based upon the nature of the data (speech or non-speech) being transmitted to the remote users.

BRIEF DESCRIPTION OF THE DRAWINGS

The features, objects, and advantages of the present invention will become more apparent from the detailed description set forth below when taken in conjunction with the drawings in which like reference characters are identified correspondingly throughout and wherein:

Figure 1 is a block diagram illustrating multiple remote (mobile) users accessing a main communications center (cell base station);

Figure 2 is a block diagram illustrating the effects of a multi-cell (multiple main communications centers) environment on data reception at a remote (mobile) user;

Figure 3 is a graph of average service quality versus number of users at a particular average transmission data rate;

Figure 4 is a graph of average service quality versus number of users for three different average transmission data rates;

Figure 5 is a flowchart of the system monitor and control operation;

Figure 6 is a communication resource pie chart for forward link communications;

Figure 7 is a communication resource pie chart for reverse link communications;

Figure 8 is a communication resource pie chart illustrating the actions to be taken with respect to different fractions of resource usage;

Figure 9 is a communication resource pie chart illustrating conditions under which the data rate would be decreased by the control mechanism of the present invention

Figure 10 is a communication resource pie chart illustrating the effects of decreasing the data rate of the previous communications resource;

Figure 11 is a block diagram of the monitor and control system for controlling reverse link communications located at the main communications center;

Figure 12 is a block diagram of the monitor and control system for controlling reverse link communications located at the remote user; and

Figure 13 is a block diagram of the forward link monitor and control apparatus.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

Figure 1 illustrates the multi-user communications system communications between remotes users 4 and the main transmission center 2. In the exemplary embodiment these communications are conducted by means of a code division multiple access (CDMA) multi- user scheme, which is detailed in U.S. Patent Serial No. 4,901,307 entitled "Spread Spectrum Multiple Access Communication System Using Satellite of Terrestrial Repeaters (CDMA)," and U.S. Patent Serial No. 5,103,459 entitled "System and Method for Generating Signal Waveform in a CDMA Cellular Telephone System (CDMA)," both assigned to the assignee of the present invention and incorporated by reference herein. The communications that occur from the remote users to the main transmission center are referred to as reverse link communications. The communications link that enables communications from remote users 4 to a cell base station 2 is referred to as the reverse link. In a CDMA system, system user capacity is a function of the level of interference in the system.

Figure 2 illustrates the two main issues that result in the need for the control of the data rate to reduce interference and increase capacity. In the exemplary embodiment of a CDMA multi-cell cellular communications network, the main capacity limit on forward link communications is the interference from neighboring cells as illustrated by the propagation lines drawn from the cell base stations 12 and the single remote user or mobile station 10. The second effect on forward link capacity in the present embodiment is illustrated by the second propagation path 18 from a single cell base station to a mobile station 10. The cause of this effect, known as multipath, is reflection off of obstruction 16 which may take the form of a

building, a mountain, or any other object that is capable of reflecting electromagnetic waves.

In the exemplary embodiment, interference is received by remote user 10 from cell base stations 12 which are not communicating with the remote user, and interference is received by multipath signals from obstruction 16. In the exemplary embodiment, the operation of a group of cells is overseen by the system controller 14 that provides the data to and from a public telephone switching network (not shown). These communications are referred to as forward link communications.

In systems like time division multiple access (TDMA) and frequency division multiple access (FDMA), a "hard" capacity limit exists due to the finite number of time slot or frequency sub-band divisions, respectively. When all of the time slots or subbands are allocated to users, the "hard" capacity limit is reached and service to any additional user is impossible. Though the users that have accessed the system before the capacity limit remain unaffected by any excluded users, the average quality of service to all users drops beyond the capacity limit since the quality of service for each additional user denied service is zero.

In multiple access schemes such as code division multiple access (CDMA) and random access systems like ALOHA and slotted ALOHA systems, a "soft" capacity limit exists. For these types of multiple access systems, the increase of the number of system users beyond a capacity limit causes a decrease in the quality of service to all users of the system. In a CDMA system, the transmissions of each user are seen as interference, or noise, to each other user. Beyond the soft capacity limit of a CDMA system, the noise floor becomes large enough to cause the predetermined allowable BER or FER to be exceeded. In random access schemes, each additional user increases the probability of a message collision. Beyond a capacity limit the message collisions grow so frequent that the need for retransmission or the resultant lost data causes the communication quality of all users to suffer.

Figure 3 is a graph of the average quality of service to users of such a multiple access communication system versus the number of users occupying the system, given a specified average data rate for all users. The average quality (Q_{ave}) of service is defined as:

$$Q_{ave} = \frac{1}{N}\sum_{i=1}^{N} Q_i \quad (1)$$

where Q_i is quality of service to user i and N is the number of users on the system.

Figure 3 also illustrates a quality line above which the average service quality is acceptable and below which the service quality is unacceptable. The intersection of the quality line with the plot of quality versus number of users curve defines the capacity limit of the system at the data rate of the system. In the exemplary embodiment of a CDMA system, messages are transmitted in 20 ms frames, and a tolerable frame error rate of 1% dictates the position of the quality line in the exemplary embodiment. It is understood that different frame sizes and error rates are equally applicable to the present invention.

Figure 4 illustrates three plots 20, 22, and 24 of average quality of service versus number users for three progressively decreasing average data rates. Plot 20 corresponds to the quality curve for a high average data rate, plot 22 corresponds to the quality curve for a moderate average data rate, and plot 24 corresponds to the quality curve for a low average data rate.

The first important feature in the plots is that the intersection of the plots with the vertical axis is progressively lower for lower link data rates. Below capacity limits, higher allowable data rates correspond to higher quality, since a high data rate allows more precise quantization of the parameters in the variable rate speech coder, resulting in cleaner sounding speech.

The second important feature in the plots is the intersections of the quality line with the three plots. The intersections of the quality line with each of the curves 20, 22 and 24 provides the capacity limit for the system at the respective data rates of curves 20, 22 and 24. The system capacities labeled CAP A, CAP B, and CAP C are the number of users that can access the system at the data rates of each of curves 20, 22 and 24. The capacity limit at a given data rate is obtained by dropping a vertical line, as shown in the diagram, from the intersection of the plot and the quality line to the horizontal axis representing the number of users. The capacity of the system increases for a fixed quality level as the data rate decreases.

Figure 5 is a flowchart illustrating the method of maximizing the average quality by controlling the data rate of transmission on the system. At block 30 the amount of communications resource that is in use is determined, based on the number of users accessing the system on the given link and the data rate transmitted by each user. The usage value computed in block 30 is passed to block 32. In block 32 the usage value is compared against a lower threshold. If the usage value is below the lower threshold

then the operation goes to block 34 where it is determined if the link is operating at a predetermined data rate maximum. If the system is operating at the predetermined data rate maximum, the operation moves to block 38 and no action is taken. If the system is operating below the predetermined data rate maximum, the operation proceeds to block 36 and the link data rate is increased.

If back at block 32 it is determined that the link usage is not too low, the operation proceeds to block 40 where the usage is compared against an upper threshold. If in block 40 the link usage is determined to be below the upper threshold, the operation proceeds to block 41 and no action is taken. If on the other hand, the link usage exceeds the upper threshold in block 40, the operation proceeds to block 42. In block 42, the system data rate is compared against a predetermined minimum. If the system data rate is greater than this predetermined minimum then the operation proceeds to block 44 where the link data rate is decreased.

If at block 42 the link data rate was determined to be equal to the minimum link data rate then the operation proceeds to block 46. At block 46 the system compares the usage to a predetermined usage maximum. If the communications resource is exhausted, that is the usage is equal to the predetermined maximum, then the operation proceeds to block 48 and access by any additional users is blocked. If the usage is below the predetermined usage maximum then, then operation proceeds to block 50 and no action is taken.

In TDMA systems, data rates can be modified by spreading data of a given user among a plurality of allocated time slots or combining the data of a plurality of users with selected ones of allocated time slots. In an alternative implementation variable data rates could be achieved in a TDMA system by allocating time slots of varying length to different users. Similarly, in FDMA systems data rates can be modified by spreading data of a given user among a plurality of allocated frequency sub-bands or combining the data of a plurality of users with selected ones of allocated frequency sub-bands. In an alternative implementation variable data rates in a FDMA system could be achieved by allocating varying frequency sub-bands sizes to different users.

In random access systems the probability of message collisions is proportional to the amount of information each user needs to send. Therefore, the data rate can be adjusted directly by sending varying size packets of data or by sending the packets at varying time intervals between transmission.

In the exemplary embodiment using a CDMA system, the amount of data necessary for transmission of speech is adjusted by use of a variable rate vocoder as detailed in Application Serial No. 08/004,484 mentioned above. The variable rate vocoder of the exemplary embodiment, provides data at full rate, half rate, quarter rate and eighth rate corresponding to 8Kbps, 4Kbps, 2Kbps and 1Kbps, but essentially any maximum average data rate can be attained by combining data rates. For example, a maximum average rate of 7Kbps can be attained by forcing the vocoder to go to half rate every fourth consecutive full rate frame. In the exemplary embodiment, the varying size speech data packet, is segmented and segments are provided at randomized times as is detailed in U.S. Patent Application Serial No. 07/846,312 entitled "Data Burst Randomizer," assigned to the assignee of the present invention and incorporated by reference herein.

A useful way of looking at the issue of communications resource capacity is to view the available communications resource as a pie chart, where the whole pie represents the complete exhaustion of the communication resource. In this representation sectors of the pie chart represent fractions of the resource allocated to users, system overhead, and unused resource.

In a TDMA or FDMA system the whole of the pie chart may represent the number of available time slots or frequency sub-bands in a given allocation strategy. In a random access system, the whole of the pie chart may represent the message rate that is acceptable before message collisions grow so great as to make the transmission link unacceptable. In the exemplary embodiment of a CDMA system, the whole of the pie chart represents the maximum tolerable noise floor wherein the overhead and signal from all other users appear as noise in the reception of the message data to and from the remoter users. In any system configuration, referring back to Figure 3, the whole of the resource pie represents the intersection of the quality line with the average quality versus number of users plot.

Figure 6 illustrates an example of a general forward link capacity pie chart. The first sector of the resource pie labeled OVERHEAD represents the portion of the transmission signal that does not carry message information. The OVERHEAD fraction of the pie represents the transmission of non-message non-user-specific data and in the exemplary embodiment is a fixed fraction of the communication resource though in other systems this overhead may vary with the number of users or other factors. The OVERHEAD may include base station identification information, timing information and base station setup information among other things. The

OVERHEAD may include pilot channel usage of the communications resource. An example of a pilot channel is detailed in U.S. Patent Serial No. 5,103,459, entitled "System and Method for Generating Signal Waveforms in a CDMA Cellular Telephone System (CDMA)," assigned to the assignee of the present invention and incorporated herein by reference. Each of the following sectors numbered 1-20 represents a the message information directed to a particular user, where the users are numbered 1-20. The last sector of the pie, moving in a clockwise direction, is labeled with a B. The sector labeled with a B represents the remaining fraction of available communication resource before unacceptable link degradation occurs.

Figure 7 is a resource pie chart for the reverse link communications. This pie chart represents the information received at the main transmission center or base station from the remote users. The only significant difference between this pie chart and the previous pie chart is in the reverse link there is no fixed OVERHEAD resource. It should also be noted that in the preferred embodiment each user uses the same fraction of communication resource in order to maximize the quality of service to all users. The method and apparatus for maintaining the condition wherein all users use the same fraction of received communication resource is detailed in U.S. Patent No. 5,056,109 entitled "Method and Apparatus for Controlling Transmission Power in a CDMA Cellular Telephone System" assigned to the assignee of the present invention and incorporated by reference herein. In this approach, each remote user transmits at a power level such that it is received at the base station as all other remote users. Preferably, each remote user transmits at a minimum power level necessary to insure a quality communication link with a base station.

Figure 8 is an action pie chart that represents the actions to be followed with respect to the resource pie charts. Labeled on the pie chart of Figure 7 are three points, a point marked INCREASE RATE, a point marked DECREASE RATE and a point marked BLOCK ADDITIONAL USERS. If the fraction of the resource pie for a given link exceeds the point marked DECREASE RATE, the transmission rate on that link should be decreased to improve the quality of service to the users. For example, if the data rate corresponding to plot 20 in figure 4 was being transmitted by all users and the number of users became greater than CAP A, the data rate would be decreased, and the system would then operate on plot 22 in figure 4. If the fraction of the resource pie for a given link falls below the point marked INCREASE RATE, the transmission rate on that link should be increased to

improve the quality of service to the users. For example, if the data rate corresponding to plot 22 in figure 4 was being transmitted by all users and the number of users dropped below CAP A, the data rate would be increased and the system would operate on plot 20 in figure 4. If the pie reaches the point marked BLOCK ADDITIONAL USERS then any additional users should be blocked from accessing the system. Note that the only way the system would reach the BLOCK ADDITIONAL USERS point is by going through the DECREASE RATE point which implies that the rate could not be further decreased.

Figures 9 and 10 illustrate the effects of decreasing the transmission rate on the resource allocation. In Figure 8, the addition of user 20 has caused the resource allocation to surpass the point at which the transmission rate should be decreased. At this point the transmission rate is decreased and the resource pie for the same users looks like Figure 9. Notice the unused portion of the resource pie labeled B is large enough to permit additional users to access the communication resource. Thus, additional users can access the communication system until the system requires the transmission rate to be decreased again. This process will continue until the rate is at a minimum. If this occurs, the system allows the pie to fill entirely and then any new users are prevented from accessing the system.

In contrast as users leave the communication resource then the fraction of the communication resource that is used decreases below the INCREASE RATE point and the system will increase the transmission rate. This can continue until the transmission rate is increased to a maximum rate or until no users are accessing the communication resource.

Figure 11 illustrates a block diagram for the monitor and control of the reverse link communication resource usage at the main communications center, which may include the cell base station and the system controller. The signals from the remote users are received at receive antenna 60. The received signals are provided to receiver 62 which provides the data in analog or digital form to energy computation element 66 and demodulators 64. The computed energy value from energy computation element 66 is provided to rate control logic 68 which compares the received signal energy to a series of thresholds. In response to the comparisons, rate control logic 68 provides a rate control signal to microprocessors 70 when the signal energy is above an upper threshold or is below a lower threshold. In other embodiments, the rate control logic 68 could also be responsive to external factors which may affect the

performance of the communications channel, such as weather conditions, etc.

The received signal from receiver 62 are provided to demodulators 64, where it is demodulated and the data for a specific user is extracted and provided to the corresponding microprocessor 70. In the exemplary embodiment, as detailed in U.S. Patent Application Serial No. 07/433,031 entitled "Method and System for Providing a Soft Handoff in Communication in a CDMA Cellular Telephone System" assigned to the assignee of the present invention and incorporated by reference herein, the received data is provided by microprocessors 70 to selector cards (not shown) in a system controller 14 that selects a best received data from received data from a plurality of main communication centers (cells), each of which contains a receiver 62 and a demodulator 64, and decodes the best received data using a vocoder (not shown). The reconstructed speech is then provided to a public telephone switching network (not shown).

In addition, microprocessors 70 receive data for forward link transmission from the vocoders (not shown) through the data interface. Microprocessors 70 combine the reverse link rate control signal, when present, with the outgoing forward link data to provide composite data packets to modulators 72. In a preferred embodiment, ones of microprocessors 70 would selectively combine the reverse link rate control when present to with outgoing forward link data. In the preferred embodiment, ones microprocessors 70 are responsive to a signal indicative of overriding conditions where upon the reverse rate control signal is not combined with the outgoing forward link data. In alternate embodiment, certain ones of said microprocessors 70 would not be responsive to the reverse link rate control signal. Modulators 72 modulate the data packets and provide the modulated signals to summer 74. Summer 74 sums the modulated data and provides it to transmitter 76 where amplified and provided to transmission antenna 78.

Figure 12 illustrates a block diagram of the remote user apparatus of the present invention for responding to the rate control signal provided in the exemplary embodiment by main transmission center 2 in figure 1. On the receive path, the signal that comprises encoded speech data and/or signaling data is received at antenna 90, which also serves as the transmission antenna by means of duplexer 92. The received signal is passed through duplexer 92 to demodulator 96. The signal is then demodulated and provided to microprocessor 98. Microprocessor 98 then decodes the signal and passes the speech data and any rate control data that

is sent by the base station to the variable rate vocoder 100. Variable rate Vocoder 100 then decodes the encoded packet of speech data provided from microprocessor 98 and provides the decoded speech data to codec 102. Codec 102 converts the digital speech signal into analog form and provides the analog signal to speaker 106 for playback.

On the transmit path of the remote user, a speech signal is provided through microphone 106 to codec 102. Codec 102 provides a digital representation of the speech signal to the variable rate vocoder 100 which encodes the speech signal at a rate determined in the exemplary embodiment in accordance with the speech activity and the received rate signal. This encoded speech data is then provided to microprocessor 98.

In the exemplary embodiment, the rate control signal is a binary signal indicating to the remote user to increase or decrease the maximum data rate. This adjustment of the data rate is done in discrete levels. In the exemplary embodiment, the remote user will increase or decrease its maximum transmission rate by 1000 bps upon receipt rate control signaling from the cell base station. In practice, this reduces the overall average data rate by 400 to 500 bps, since the vocoder is only encoding the speech at the maximum rate 40-50% of the time in a normal two-way conversation. In the exemplary embodiment, the silence between words is always encoded at the lower data rates.

For example, if the remote user is currently operating with a maximum transmission data rate of full rate or rate 1 (8 Kbps), and a signal decrease its maximum data rate is received, the maximum transmission data rate will be decreased to 7/8 (7 Kbps) by forcing every fourth consecutive full rate frame of data to be encoded at half rate (4Kbps). If on the other hand, the remote user is operating under control of the cell base station at a maximum transmission rate of 3/4 (6 Kbps) and the cell base station signals the remote user to increase its maximum data rate, then the remote user will use a rate 7/8 (7 Kbps) as a maximum transmission data rate. In a simplified embodiment the rates could simply be limited to one of the discrete rates provided by variable rate vocoder 100 (i.e.. rates 1, 1/2, 1/4 and 1/8).

Microprocessor 98, also, receives non-speech data that can include signaling data or secondary data such as facsimile, modem, or other digital data that needs communication to the cell base station. If the digital data being transmitted by the remote user is of a form not conducive to variable rate transmission (i.e. some facsimile or modem data) then microprocessor

98 can decide based upon the service option of the remote user whether to vary the transmission rate in response to the rate control signal.

Modulator 108 modulates the data signal and provides the modulated signal to transmitter 110 where it is amplified and provide through duplexer 92 to antenna 90 and transmitted over the air to the base station. It is also envisioned in the present invention that the remote user could monitor the reverse link communication resource and respond in an open loop manner to adjust its transmission rate.

Figure 13 illustrates a block diagram of an exemplary forward link rate control apparatus. Speech data is provided to vocoders 120 where the speech data is encoded at a variable rate. In the present invention the encoding rate for the speech data is determined in accordance with the speech activity and a rate control signal when present. The encoded speech is then provided to microprocessors 122, which also may receive non-speech data from an external source (not shown). This non-speech data can include signaling data or secondary data (facsimile, modem or other digital data for transmission). Microprocessors 122 then provide data packets to modulators 124 where the data packets are modulated and provided to summer 126. Summer 126 sums the modulated signal from modulators 124 and provides the sum signal to transmitter 128 where the signal is mixed with a carrier signal, amplified and provided to antenna 130 for transmission.

The summed modulated signal from summer 126 is also provided to energy computation unit 132. Energy computation unit 132 computes the energy of the signal from summer 126 for a fixed time period and provides this energy estimate to rate control logic 134. Rate control logic 134 compares the energy estimate to a series of thresholds, and provides a rate control signal in accordance with these comparisons. The rate control signal is provided to microprocessors 122. Microprocessors 122 provide the rate control signal to vocoders 120 for control of the maximum data rate of speech data. Optionally, microprocessors 122 can also use the rate control signal to control the data rate of non-speech data sources (not shown). the rate control signal can be provided selectively to ones of microprocessors 122 or alternately selects ones of microprocessors 122 can be responsive to a globally provided rate control signal.

The open loop form of control on the forward link described above can also operate in a closed loop, which can be responsive to signals from the remote stations indicative of capacity limits being reached, such as high frame error rates or other measurable quantities. Rate control logic 134 can

be responsive to external interferences of various kinds which may also affect the performance of the communications channel.

The previous description of the preferred embodiments is provided to enable any person skilled in the art to make or use the present invention. The various modifications to these embodiments will be readily apparent to those skilled in the art, and the generic principles defined herein may be applied to other embodiments without the use of the inventive faculty. Thus, the present invention is not intended to be limited to the embodiments shown herein but is to be accorded the widest scope consistent with the principles and novel features disclosed herein.

WE CLAIM:

CLAIMS

1. In a communications network wherein a plurality of remote users each having a transmitter communicate message signals to a communications center having a receiver, a subsystem for optimizing communications quality in accordance with system usage and capacity comprising:

monitor means for determining said system usage and conditionally providing a rate control signal in accordance with said usage level; and

a plurality of response means each collocated with a corresponding one of said remote users for receiving said rate control signal and adjusting said data rate of said corresponding one of said remote users in accordance with.

2. The subsystem of Claim 1 wherein said monitor means is collocated with said transmission center, said subsystem further comprising:

communications center transmitter means for transmitting messages to said remote users and for transmitting said rate control signal to said remote users; and

plurality of remote receiver means, each of said plurality of remote receiver collocated with a corresponding one of said remote users for receiving said rate control signal and for providing said rate control signal to a corresponding one of said response means.

3. The subsystem of Claim 1 wherein said monitor means determines said system usage by measuring the energy of said message signals for a predetermined time period.

4. The subsystem of Claim 1 wherein said response means comprises:

processor means for receiving said rate control signal and providing a rate command signals in response to said rate control signal; and

variable rate vocoder means for receiving speech data and said rate command signals and encoding said speech data at a rate in accordance with said command signals.

5. The subsystem of Claim 4 wherein said variable rate vocoder means further encodes said speech data in accordance with the energy of said speech data.

6. The subsystem of Claim 4 wherein said processor means is further for receiving non-speech data for transmission and for providing said non-speech data at a rate in accordance with said rate control signal.

7. A variable rate transceiver comprising:

a receiver for receiving a signal comprising message data and a rate control command;

a variable rate vocoder for receiving speech data and encoding said speech data in accordance with said rate control command; and

a transmitter for transmitting said encoded speech data.

8. The variable rate transceiver of Claim 7 further comprising:

a demodulator disposed between said receiver and said variable rate vocoder for demodulating said received signal; and

a processor disposed between demodulator and said variable rate vocoder for receiving said demodulated signal and separately providing said message data and said rate control command.

9. The variable rate transceiver of Claim 8 wherein said processor is further for receiving non-speech data for transmission.

10. The variable rate transceiver of Claim 7 further comprising a modulator disposed between said variable rate vocoder and said transmitter for modulating said encoded speech data.

11. The variable rate transceiver of Claim 7 further comprising a modulator disposed between said variable rate vocoder and said transmitter for modulating said encoded speech data.

12. At a base station, an apparatus for controlling the user capacity of said base station comprising:

usage determination means for measuring usage of said base station;

rate control means for comparing said measured usage against at least one predetermined value and selectively providing a rate control signal in accordance with said comparisons; and

transmitter means for transmitting said rate control signal.

13. The apparatus of Claim 12 further comprising processor means for receiving message data for transmission to said remote users and said rate control signal and combining said message data with said rate control signal to provide a composite data packet.

14. The apparatus of Claim 13 further comprising a modulator means disposed between said processor means and transmitter for modulating said composite data packet.

15. In a communication system wherein a base station communicates messages on a forward link with a plurality of remote users an apparatus of controlling the data rate of said message communications, comprising:

usage determination means for determining a usage value of said forward link;

rate control logic means for receiving said usage value, comparing said usage value to at least one predetermined threshold value and conditionally providing a rate control signal in accordance with said comparisons; and

at least one variable rate data source for providing data at a rate in accordance with said rate control signal.

16. The apparatus of Claim 15 wherein said at least one variable rate data source comprises at least one variable rate vocoder means for encoding speech data at variable rates.

17. The apparatus of Claim 15 wherein said usage determination means measures the energy of a signal for transmission to said remote users.

18. A method for optimizing usage of a communications resource, comprising the steps of:

measuring said usage of said communications resource;

comparing said measured usage against at least one predetermined threshold; and

adjusting data rates of communications on said communications resource in accordance with said comparisons.

19. The method of Claim 18 wherein said step of comparing said measured usage against at least one predetermined threshold comprises comparing said usage against a predetermined high usage threshold, and wherein said step of adjusting data rates of communications on said communications resource comprises decreasing the data rate of communications when said usage exceeds said high usage threshold.

20. The method of Claim 18 wherein said step of comparing said measured usage against at least one predetermined threshold comprises comparing said usage against a predetermined low usage threshold, and wherein said step of adjusting data rates of communications on said communications resource comprises increasing the data rate of communications when said usage falls below said low usage threshold.

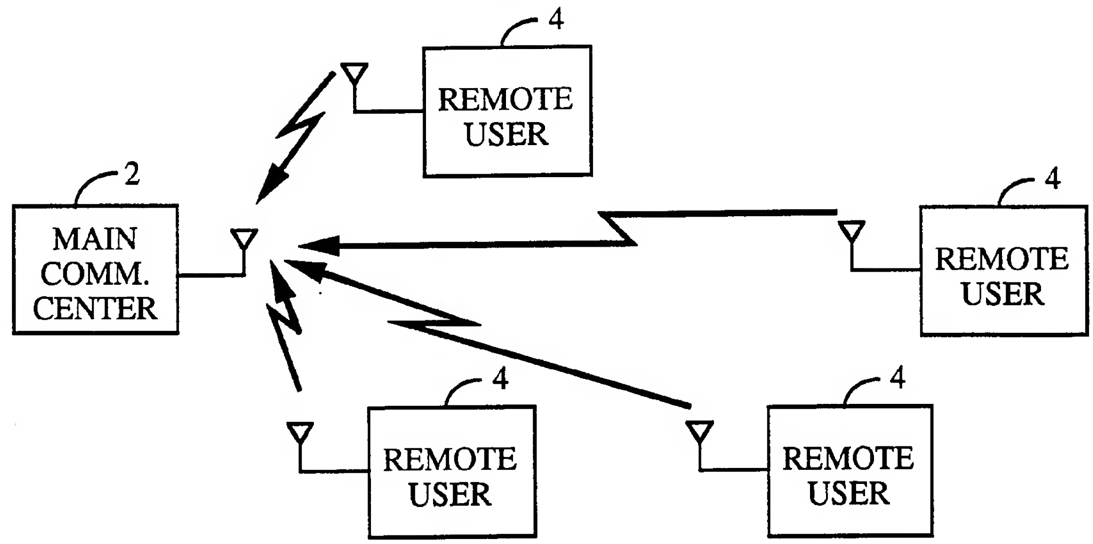
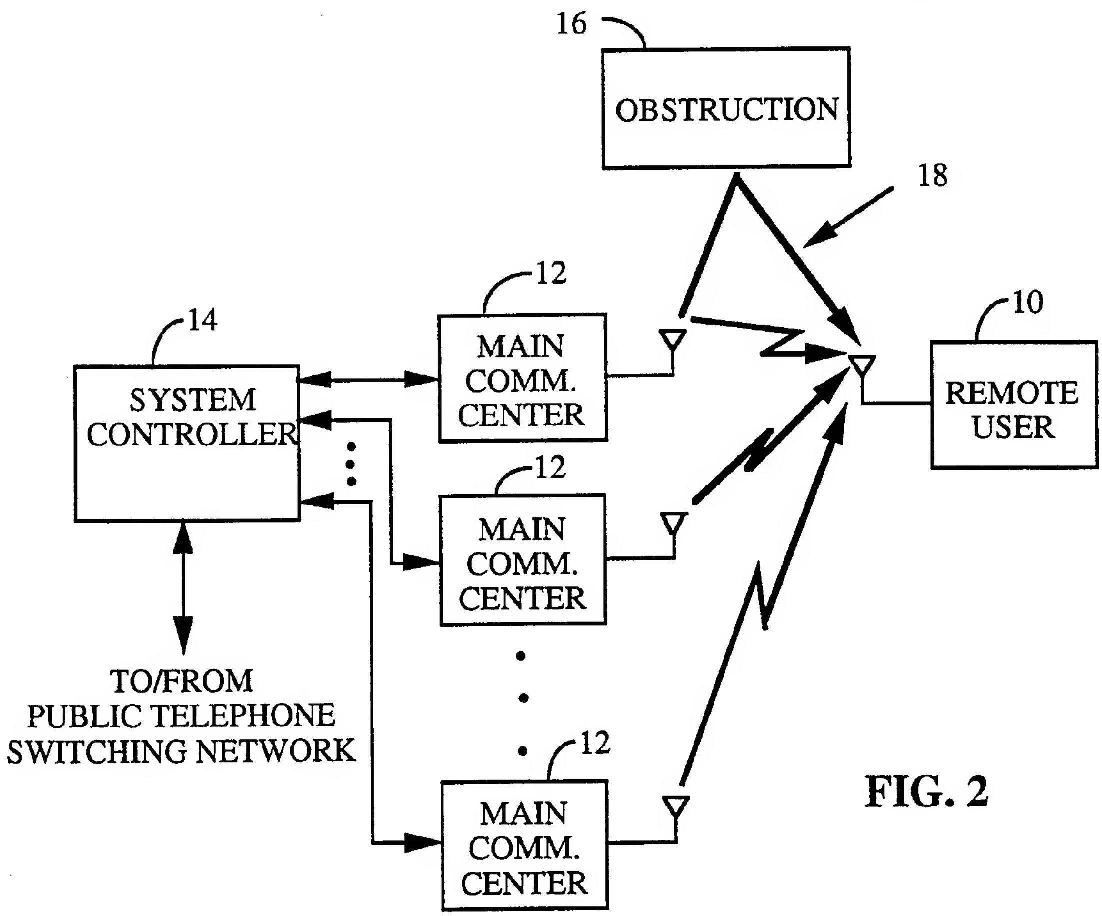

4
REMOTE USER
2
MAIN COMM. CENTER
4
REMOTE USER
4
REMOTE USER
4
REMOTE USER

FIG. 1


16
OBSTRUCTION
18
12
14
10
SYSTEM CONTROLLER
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REMOTE USER
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MAIN COMM. CENTER
TO/FROM PUBLIC TELEPHONE SWITCHING NETWORK
12
MAIN COMM. CENTER

FIG. 2

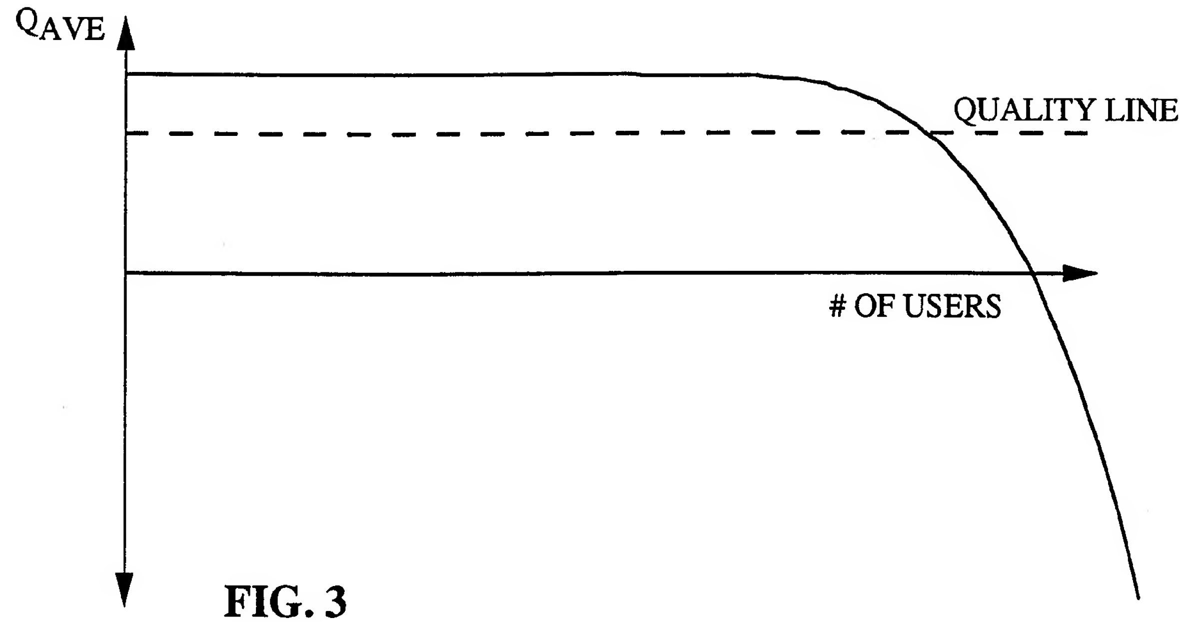

Q_{AVE}
QUALITY LINE
OF USERS

FIG. 3

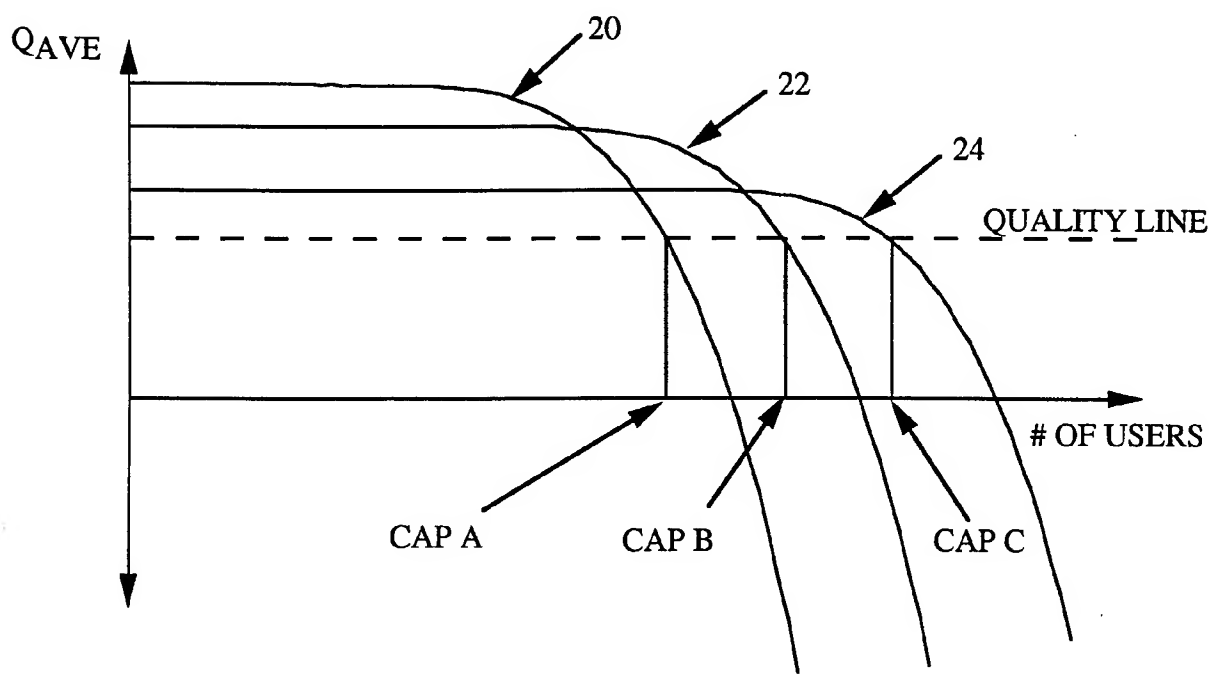
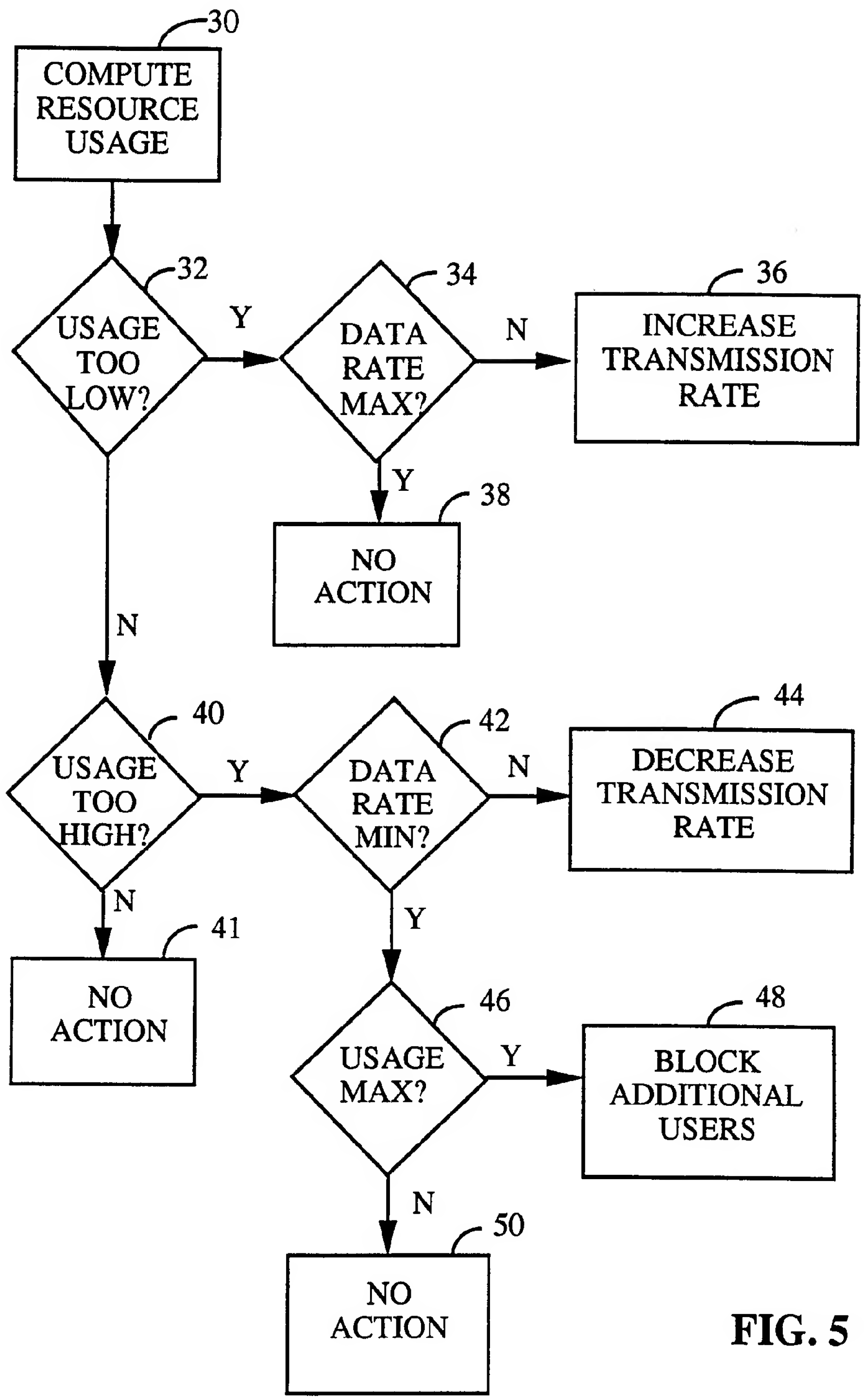

Q_{AVE}
20
22
24
QUALITY LINE
OF USERS
CAP A
CAP B
CAP C

FIG. 4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32
USAGE TOO LOW?
Y
34
DATA RATE MAX?
N
36
INCREASE TRANSMISSION RATE
Y
38
NO ACTION
N
40
USAGE TOO HIGH?
Y
42
DATA RATE MIN?
N
44
DECREASE TRANSMISSION RATE
N
41
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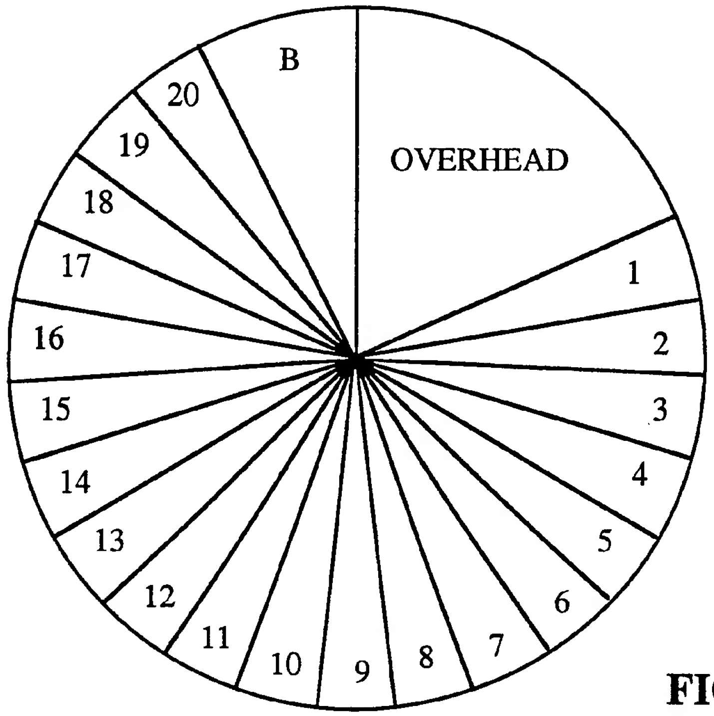
Y
46
USAGE MAX?
Y
48
BLOCK ADDITIONAL USERS
N
50
NO ACTION

FIG. 5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OVERHEAD
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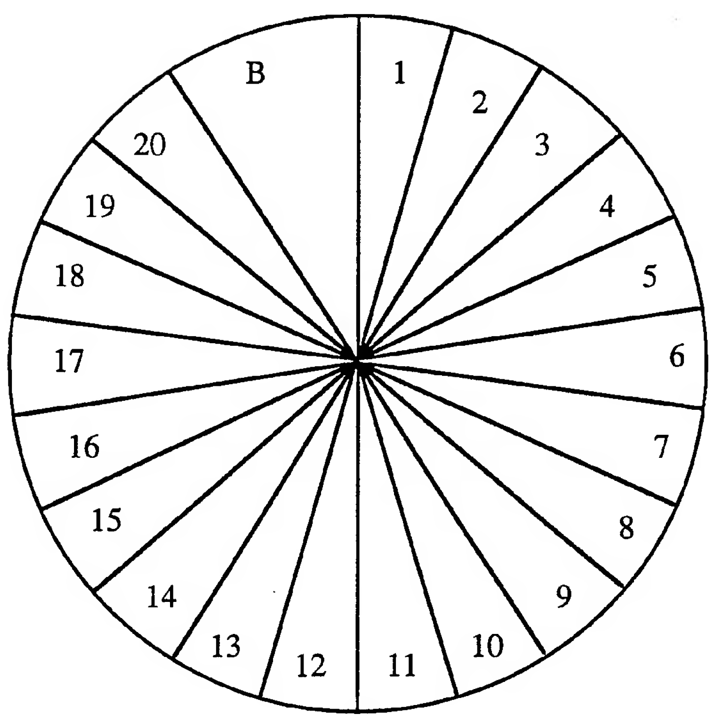
FIG. 6


B
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20

FIG. 7

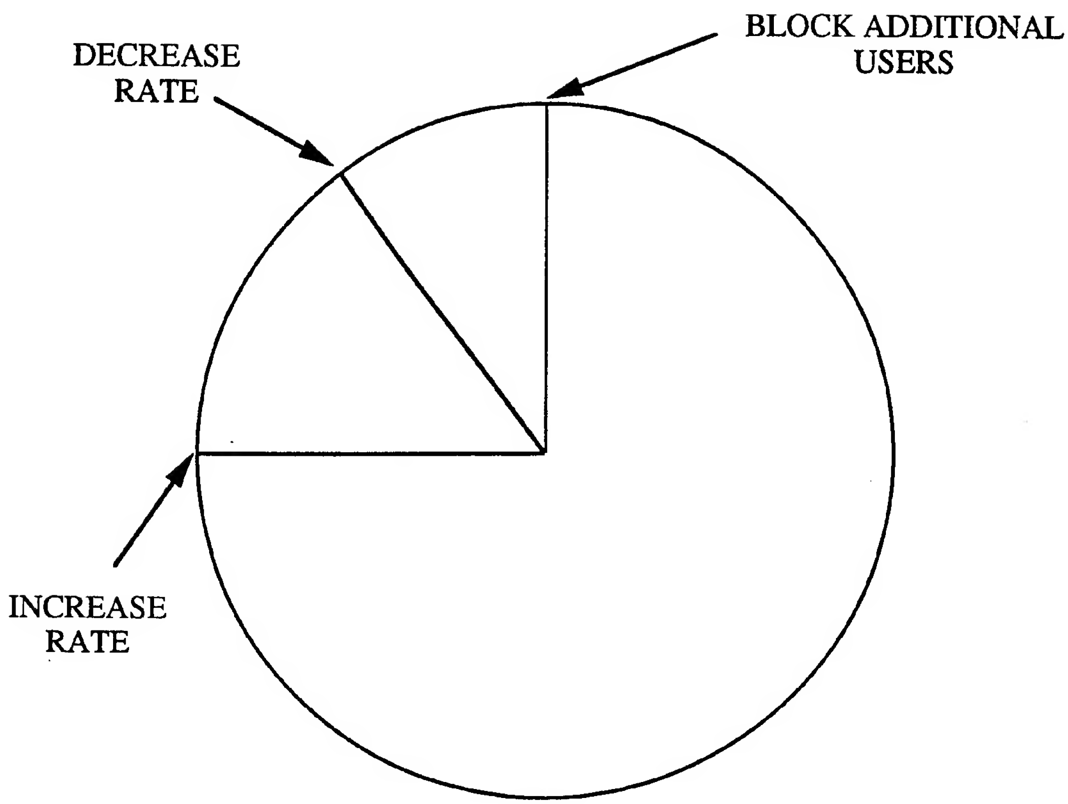
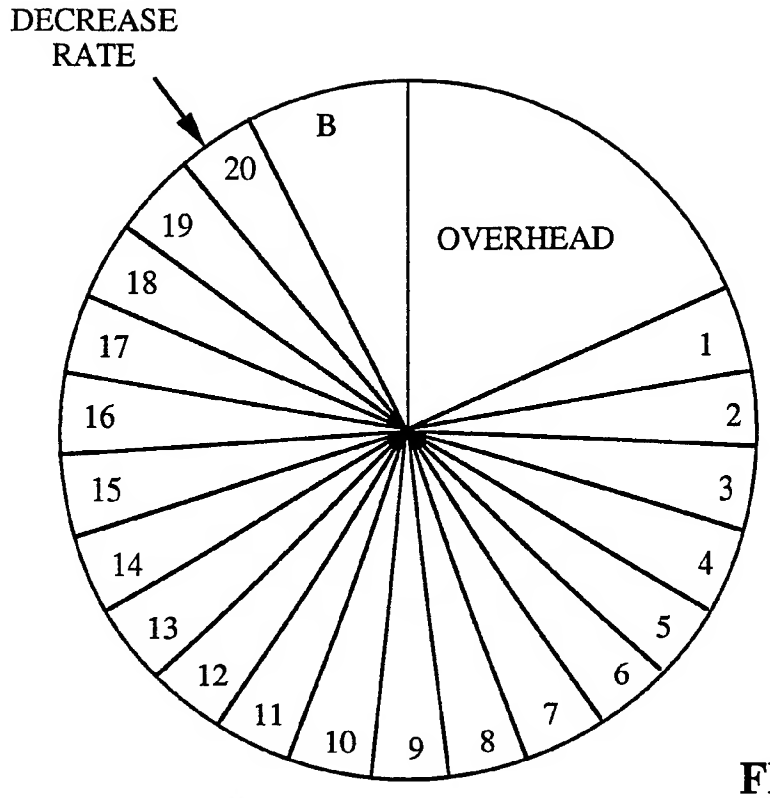


FIG. 8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OVERHEAD
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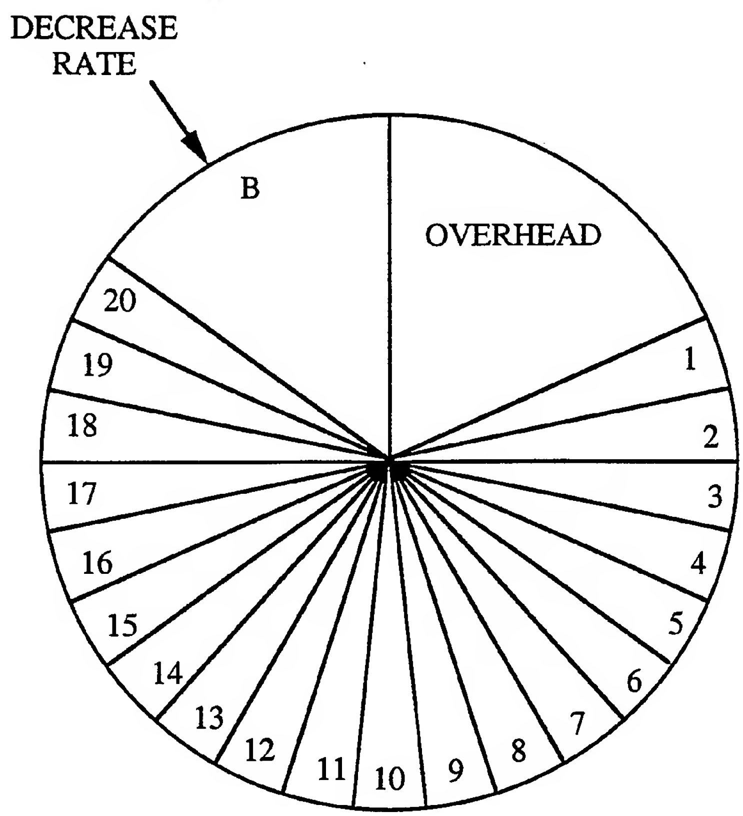
20

FIG. 9


DECREASE
RATE
B
OVERHEAD
1
2
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8
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10
11
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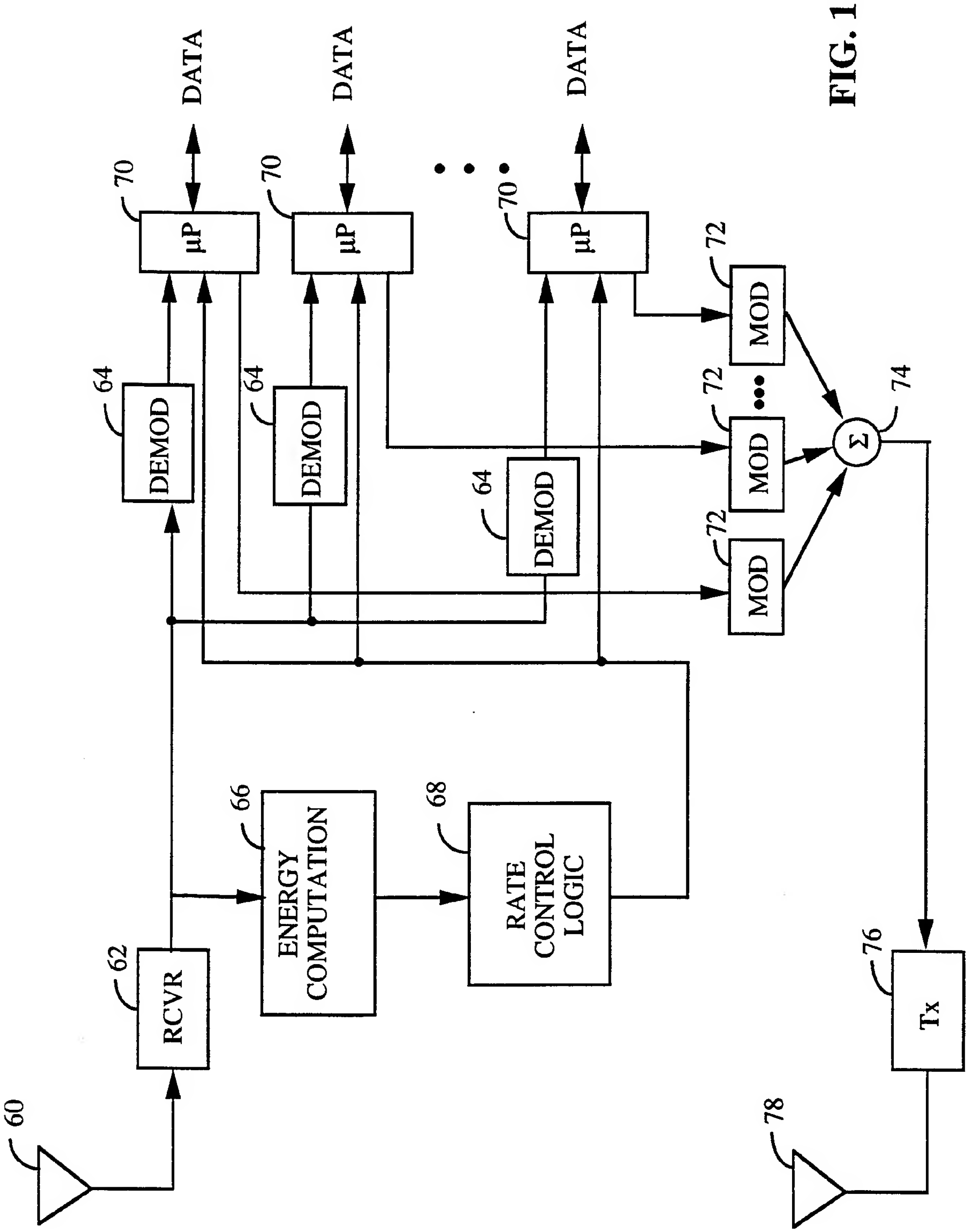
19
20

FIG. 10


DATA
DATA
DATA
70
70
70
μP
μP
μP
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64
64
DEMOD
DEMOD
DEMOD
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72
72
MOD
MOD
MOD
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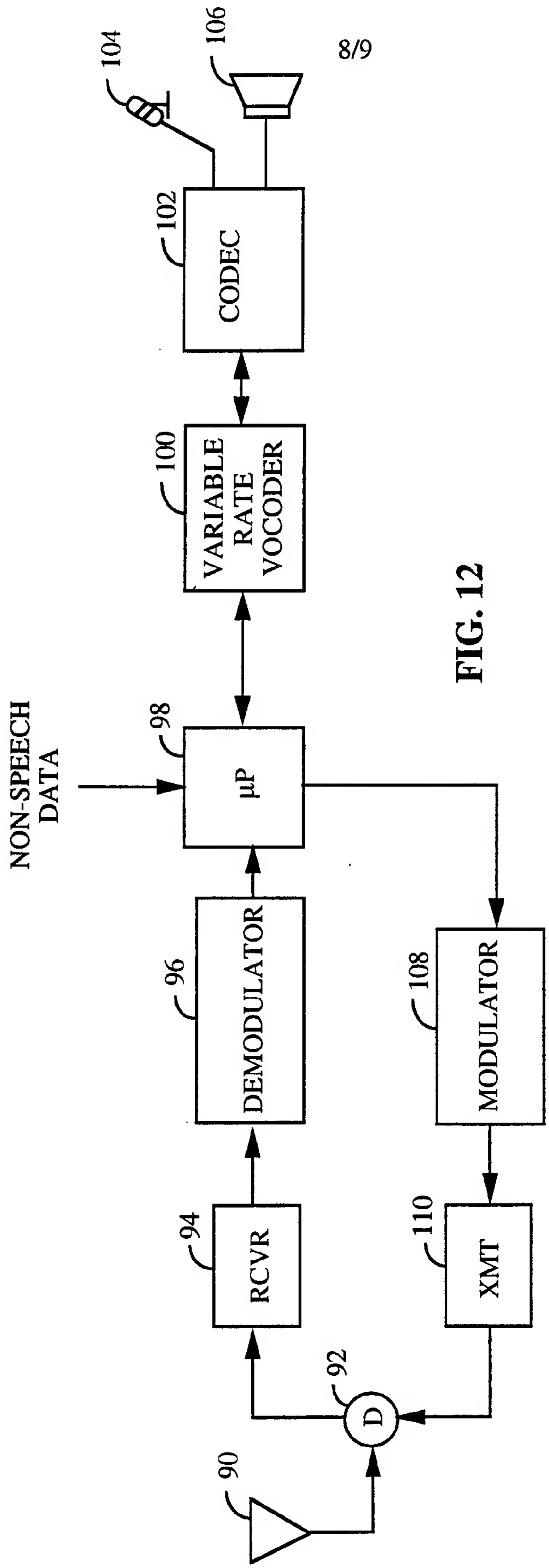
Σ
60
62
RCVR
66
ENERGY COMPUTATION
68
RATE CONTROL LOGIC
76
Tx
78

FIG. 11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100
VARIABLE RATE VOCODER
NON-SPEECH DATA
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μP
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DEMODULATOR
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MODULATOR
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RCVR
110
XMT
92
D
90

FIG. 12

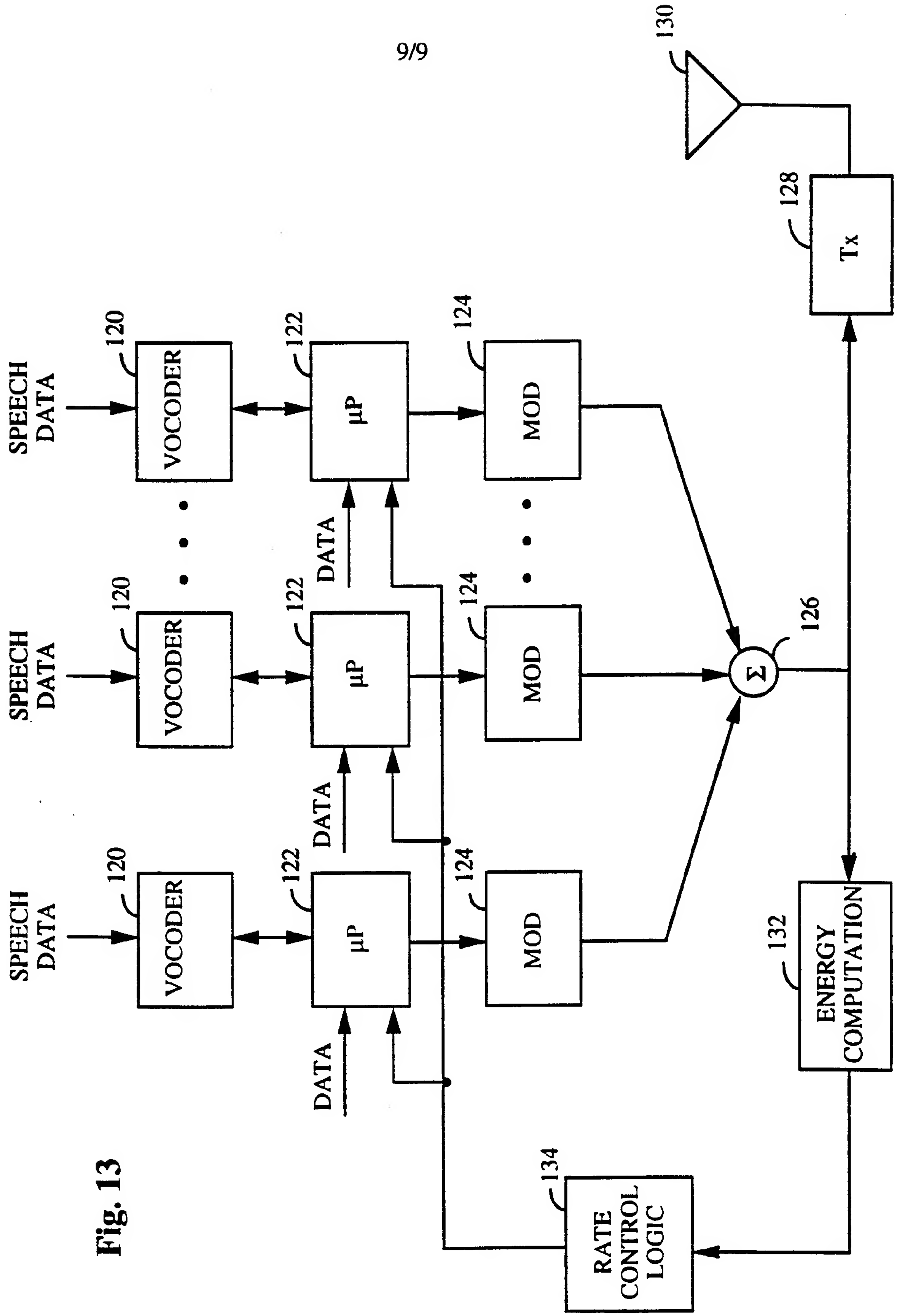
SPEECH DATA
VOCODER
µP
MOD
DATA
Σ
Tx
ENERGY COMPUTATION
RATE CONTROL LOGIC
120
122
124
126
128
130
132
134

Fig. 13

INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 94/10087

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 H04B7/26 H04L1/12 H04Q7/38

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 H04Q H04B H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	EP,A,0 538 546 (MOTOROLA) 28 April 1993 see column 2, line 58 - line 21	1-20
X	EP,A,0 353 759 (NORAND CORPORATION) 7 February 1990 see column 2, line 29 - column 3, line 1	1,2,12, 13,15,18
X	EP,A,0 472 511 (ERICSSON) 26 February 1992 see column 3, line 37 - line 57	1,2,12, 13,15,18

☐ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

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EP-A-0538546	28-04-93	WO-A-	8706082	08-10-87
		AU-A-	5589086	20-10-87
		DE-D-	3689979	25-08-94
		EP-A-	0261112	30-03-88
		EP-A-	0412583	13-02-91
EP-A-0353759	07-02-90	US-A-	4910794	20-03-90
		AU-B-	632055	17-12-92
		AU-A-	3927889	08-02-90
		CA-A-	1316218	13-04-93
		GB-A,B	2223914	18-04-90
		US-A-	5070536	03-12-91
EP-A-0472511	26-02-92	AU-B-	642760	28-10-93
		AU-A-	8261991	27-02-92
		JP-A-	4234232	21-08-92
		NZ-A-	239283	27-09-94
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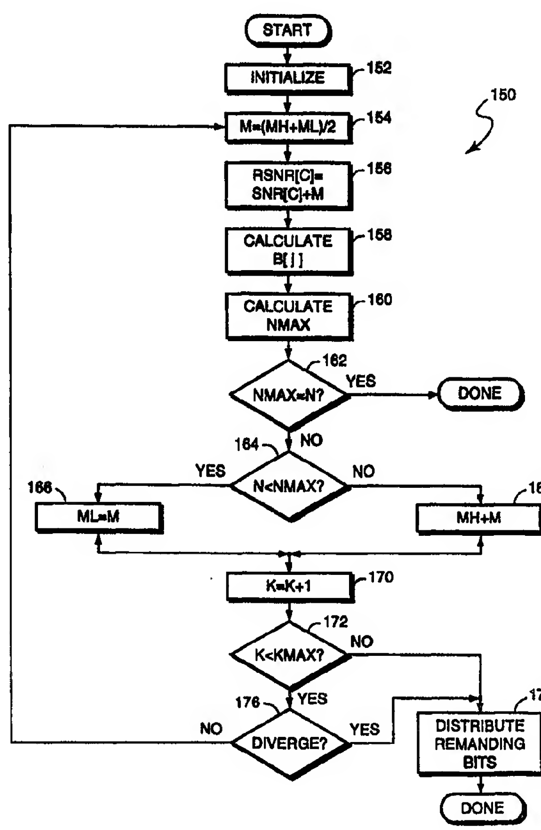
(57) Abstract

Data is distributed among the channels of an asynchronous data subscriber loop (ADSL) communications system in accordance with an adaptive algorithm which from time to time measures the signal to noise ratio of the various channels and finds a margin for each channel dependent on achievement (where possible) of a given bit error rate and a desired data transmission rate. The margin distribution is achieved by augmenting the constellation signal to noise ratio to enhance computational efficiency and allow redetermination of bit allocation tables during transmission as necessary. Pairs of bit allocation tables are maintained at the transmitter and receiver and one table of each pair at the transmitter and receiver is updated while the other pair is in use for controlling communication.



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ADAPTIVE BIT ALLOCATION FOR VARIABLE BANDWIDTH MULTICARRIER COMMUNICATION

TECHNICAL FIELD

This application relates to the field of electronic communication and more particularly to the field of multiband digital signal communication.

BACKGROUND OF THE INVENTION

Conventional multicarrier digital communication is a technique for transmitting and receiving digital signals using a plurality carriers (subchannels) having different frequencies. Each of the subchannels is used to communicate a different portion of the signal. The transmitter divides the signal into a number of components, assigns each component to a specific one of the carriers, encodes each of the carriers according to the component assigned thereto, and transmits each of the carriers. The receiver decodes each received carriers and reconstructs the signal.

The maximum amount of information that can be encoded onto a particular subcarrier is a function of the signal to noise ratio of the communication channel with respect to that subcarrier. The signal to noise ratio of a communication channel can vary according to frequency so that the maximum amount of information that can be encoded onto one carrier may be different than the maximum amount of information that can be encoded onto another carrier.

Bit loading is a technique for assigning bits to subchannels according to each subchannel's signal to noise ratio. A bit allocation algorithm provides a bit allocation table that indicates the amount of information (in bits) that is to be encoded on each of the carriers. That is, for a multicarrier communication system with J carriers, a bit allocation table B[j] indicates, for each j = 1 to J, the amount of information that is to be encoded onto each of the J carriers.

Shaping the transmission to match the channel characteristics is known. For example, a technique known as "water pouring" was introduced by Gallager in 1968

("Information Theory and Reliable Communication", page 389) and by Wozencraft in 1965 ("Principles of Communication Engineering", pp. 285-357). Water pouring involves distributing the energy of the transmission signal according to the channel frequency response curve (a plot of the signal to noise ratio as a function of frequency). The frequency response curve is inverted and the available signal energy (the "water") is "poured" into the inverted curve so that more of the energy is distributed into those portions of the channel having the highest signal to noise ratio. In a multicarrier system in which the transmission band is divided into numerous subchannels, throughput can be maximized by putting as many bits in each subcarrier as can be supported given the "water pouring" energy and a desired error rate.

Other techniques for allocating bits among carriers of a multicarrier signal are known. U.S. Patent No. 4,731,816 to Hughes-Hartogs discloses a bit loading scheme where one bit at a time is incrementally added to each subcarrier until a maximum rate is achieved. Subcarriers that require the least amount of additional power to support an additional bit are selected first.

U.S. Patent No. 5,479,477 to Chow et al. discloses a bit loading scheme that is capable of either maximizing the throughput or maximizing the margin for a particular target data rate. Unlike Hughes-Hartogs, Chow et al. determines the bit loading table one carrier at a time (rather than one bit at a time). In Chow et al., all the carriers are sorted in descending order according to the measured signal to noise ratio. The initial subchannels that are selected are the ones capable of carrying the most bits. Using the Chow et al. scheme to maximize the data rate provides a bit loading table similar to that provided by the Hughes-Hartogs algorithm.

In order for the receiver to correctly interpret the received data, both the transmitter and the receiver must use the same bit loading table. When the bit loading algorithm is performed during the initialization phase of communication, the resulting bit allocation table is communicated between the transmitter and receiver to ensure that both the transmitter and the receiver are using the same bit loading table. However, in the event that the communication channel signal to noise ratio characteristics change during communication, it may be necessary to update/change the bit allocation table to more appropriately match the transmission with the channel characteristics. However,

when the bit allocation table is changed, it is necessary to synchronize use of the new table with both the transmitter and the receiver. If the transmitter and the receiver use different bit allocation tables at any time, the communications link will suffer significant errors in those subchannels in which the bit allocation tables do not agree.

In addition, determining a new bit allocation table can be time consuming, especially if the bit loading algorithm is computationally intensive, such as that disclosed by Hughes-Hartogs where the bit allocation table is constructed one bit at a time. If the bit allocation table is to be calculated many times during communication between the transmitter and receiver, then spending a relatively long amount of time recalculating the bit allocation table (and hence not communicating data) is undesirable.

One solution is to simply not change the bit loading table after initialization. However, this may be unacceptable in cases where the communication channel signal to noise ratio changes during data transmission. Accordingly, it is desirable to be able to determine a bit loading table relatively quickly and to be able to synchronize use of the new table by the transmitter and the receiver.

SUMMARY OF THE INVENTION

In accordance with the present invention, a pair of bit allocation tables are maintained at both the transmitter and the receiver. These tables are updated as needed, using measurements of the signal to noise ratio performed on known data transmitted to the receiver in a control frame separate from the data frame. The transmitter signals the receiver as to which of the two tables is to be used for subsequent communication. Preferably, this is done by transmitting a flag from the transmitter to the receiver at some point during the data transmission; this causes the receiver to thereafter switch the bit loading table it is using for communication to synchronize with the corresponding table at the transmitter.

In the preferred embodiment of the invention, although the invention is not restricted thereto, 69 "frames" of 245.5 microseconds duration each are used to form a "superframe" of 16.94 milliseconds. The first frame of each superframe comprises a control frame that is used to transmit a standard (known) data set from the transmitter to the receiver; the remaining frames contain data. The receiver measures the signal to

noise ratios of the received data in this frame for each of the channels and uses this to calculate channel bit allocations for subsequent data transmissions. In practice, it has not been found necessary to calculate the signal to noise ratios for each and every super-frame, although this can, of course, be done. Rather, we have found it sufficient for most data transmissions to measure the signal to noise ratios of the channels over several frames, average them, update the bit allocation tables based on the resultant values, and use the bit allocations tables so determined over hundreds or thousands of subsequent frames.

The bit allocation table updating is performed by comparing the measured signal to noise ratio (SNR) in each channel with a constellation signal to noise ratio $SNR[c_j]$, that has been augmented by a trial noise margin M, $SNRa[c_j] = SNR[c_j] + M$. The constellation signal to noise ratio, $SNR[c_j]$, specifies the number of bits c_j ("constellation size") that can be transmitted over a channel j given a specific signal to noise ratio SNR_j, where c_j may vary, for example, from 1 to 15. The value of the margin M is dependent on the difference between the amount of data (i.e., number of bits) that can be transmitted across the channels in accordance with the augmented constellation signal to noise ratio $SNR_a[c_j]$ and the amount that is desired to be transmitted (the "target data rate"), N. The value of this margin is varied in order to optimize it for the particular communication conditions as manifested by the measured signal to noise ratios, SNR_j.

In particular, the total number of bits that may be transmitted over J channels, each characterized by signal to noise ratio SNRj, is $N_{max} = \sum_{j=1}^{J} c_j$, where the respective c_j are determined from the measured signal to noise ratios, SNR_j. See, for example, "Digital Communications" by John G. Proakis, pp. 278ff for channel capacity calculations for quadrature amplitude modulation (QAM) systems, the preferred form of transmission for this invention. Preferably, the channel capacity calculations are performed in advance and stored in the form of lookup tables for rapid access. In the preferred embodiment described herein, the margin M is determined as $M = (10/J)^* (N_{max} - N)$. The augmented constellation signal to noise ratio is then given by

$SNR_a[c_j] = SNR[c_j] + M$, and this value is used to determine (e.g., by table lookup as described above) the number of bits that can be transmitted over a channel. By augmenting the constellation signal to noise ratio, $SNR[c_j]$, rather than the channel signal to noise ratio, SNR_j, fewer additions are required, since the range of constellation sizes (e.g., $c_j = 1 \ldots 15$) is typically smaller than the range of channels (e.g., $j = 1 \ldots 256$).

As long as the amount of data that can be transmitted over the channels in a given interval differs (as determined by the calculations just described) from the amount of data desired to be transmitted in that interval, i.e., $N_{max} \neq N$, and assuming that certain other exit conditions have not been satisfied, the receiver cycles through a loop that repeatedly adjusts the margin M and recalculates N_{max}. To do this, the receiver sets a high margin threshold M_H and a low margin threshold M_L. During those superframes in which the bit allocation table is to be recalculated, the high threshold and low threshold margins are initialized to either a first state ($M_H = 0$, $M_L = (10/J)*[N_{max} - N]$) or a second state ($M_L = 0$, $M_H = (10/J)*[N_{max} - N]$) dependent on whether N_{max} is greater than N or less than N.

Thereafter, in each iteration, either the high or the low margin is adjusted in the search for the condition in which $N_{max} = N$. Specifically, at the beginning of subsequent (non-initialization) iterations, the margin is set to the average of the high and low margin thresholds, $M = (M_H + M_L)/2$, and the augmented constellation signal to noise ratio $SNR_a[c_j]$, the bit allocation table B[j], and the calculated capacity N_{max} are determined.

If the calculated capacity exceeds the desired capacity, i.e., $N_{max} > N$, the receiver increases the low margin threshold margin to M, i.e., it sets $M_L = M$. If the calculated capacity is less than the desired capacity, i.e., $N_{max} < N$, the receiver decreases the high threshold, i.e., it sets $M_H = M$. The iteration then repeats.

The receiver exits from the loop on the occurrence of any of several conditions. A first occurs when it is determined that $N_{max} = N$. This is the desired solution, and represents an optimum equal distribution of margin over the communication channels.

A second occurs when the test condition (N_{max} - N) is diverging. A third occurs when the desired equality is not achieved after a defined number of iterations. In one system implemented according to the preferred embodiment described herein, we have found a limit of 16 iterations sufficient.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a schematic diagram of an ADSL communications system showing bit allocation tables in accordance with the present invention;

Fig. 2 is a diagram of control and data frames as used in connection with the present invention;

Fig. 3 is a graph illustrating a multicarrier communication system.

Fig. 4 is a graph illustrating signal-to-noise ratio as a function of frequency.

Fig. 5 is a graph illustrating bit loading and margin for a multicarrier communication system.

Fig. 6 is a flow chart illustrating a bit loading algorithm for a multicarrier communication system.

Fig. 7 is a flow chart illustrating initialization for the bit loading algorithm of Fig. 6.

Fig. 8 is a flow chart illustrating operation of a receiver software for calculating, modifying, and synchronizing a change in a bit allocation table used in a multicarrier communication system.

DETAILED DESCRIPTION OF AN ILLUSTRATIVE EMBODIMENT

In Figure 1, a transmitter 10 for use in asynchronous data subscriber loop (ADSL) communications has first and second bit allocation tables 12 and 14 for use in assigning data to a plurality of channels for transmission to a remote receiver 16 which has corresponding bit allocation tables 20 and 22. The tables operate in pairs under control of a table controller 24 at the transmitter. In accordance with ADSL practice, a digital signal s(t) to be transmitted to a receiver is distributed over a plurality of channels f_1, f_2, . . . f_J, in accordance with channel allocation assignments stored in the bit allocation tables.

In particular, the tables B[j] define, for each channel j, the number of bits that can reliably be transmitted over a particular channel at a given bit error rate at the specific signal to noise ratio measured for that channel. These tables are determined as described in detail herein, and may vary from time to time during the course of a transmission.

At any given time, a single table, e.g., table 12, is used for transmission at the transmitter, and a corresponding table, e.g., table 20, is used for reception at the receiver. These tables are images of each other, i.e., contain the same data, and are used in pairs, so that reliable communication can occur. Similarly, tables 14 and 22 are images of each other and are used in pairs.

A table control unit 24 at the receiver controls the formation of the bit allocation tables 12, 14, 20, and 22. It measures the signal to noise ratio on each of the channels $f_1, f_2, \ldots f_J$, compares the measured values with predetermined values defining the bit capacity of a channel at given signal to noise values, augmented with noise margins as described herein, and thus determines the bit allocation for each channel. The allocations so defined are stored in the tables 20 and 22 at the receiver. They are also transmitted back to the transmitter, e.g., via a control channel 26, and are there stored as the tables 12 and 14, respectively. After initial loading, the transmission is advantageously arranged such that only updated tables are transmitted back to the transmitter.

At the transmitter 10, a table switch unit 28 selects which of the two table pairs (12, 20; 14, 22) are to be used in a given transmission and reception. Typically, a given pair will continue in use until the communication conditions change sufficiently that the bit allocations among the channels change. At that time, a new table must be formed at the receiver, and communicated to the transmitter. When this occurs, the table switch unit 28 typically will switch to the new table for subsequent transmissions. When it does so, it transmits a flag to the receiver that indicates that a switch to the alternative pair is to take place. This switch will usually be made effective as of the next superframe, but may, by prearrangement with the receiver, be made effective at some agreed upon point after that.

Fig. 2 is a diagram of a superframe 30. It is formed from a control frame 32 and a number of data frames 34. During the control frame interval, the transmitter sends to the receiver a known signal from which the receiver can measure the signal to noise ratio

of each of the channels in order to calculate the bit allocations. The remaining frames of the superframe comprise data frames for the transmission of the desired data. In a preferred embodiment of the invention, there are one control frame and 68 data frames, each of 245.5 microsecond duration, for a superframe time of 16.94 milliseconds.

Referring to Fig. 3, a graph 100 illustrates multicarrier signal transmission. The graph 100 has a horizontal axis 102 representing frequency wherein lower frequencies are toward the left side of the axis 102 while higher frequencies are toward the right side of the axis 102. The graph 100 illustrates that a multicarrier signal, incorporating J discrete carrier signals, is transmitted via carriers at frequencies $f_1, f_2, \ldots f_j$.

Each of the carriers shown in the graph 100 is capable of transmitting a certain number of bits of information. Accordingly, the total number of bits transmitted via the multicarrier signal is the sum of the number of bits that can be transmitted by each of the carriers. For example, if each of the carriers can transmit three bits of information, then the signal shown in the graph 100 can transmit a total of J*3 bits of information.

In a preferred embodiment, each of the carriers transmits information using quadrature amplitude modulation (QAM), a conventional digital signal encoding technique where different combinations of amplitude and phase of each carrier signal represent different digital values. For example, a carrier signal can be encoded using two different possible amplitudes (A1 and A2) and two different possible phases (P1 and P2) so that the carrier can represent one of four possible values: a first value when the carrier signal has amplitude A1 and P1, a second value corresponding to a combination A1 and P2, a third value corresponding to a combination A2 and P1, and a fourth value corresponding to a combination A2 and P2. The various combinations of amplitude and phase for a given carrier signal is called a "constellation". Note that the number of bits that can be transmitted via a particular carrier is a function of the maximum possible constellation size for that carrier.

For each carrier, the maximum size of the constellation, and hence the maximum number of bits that can be transmitted via that carrier, is a function of the signal to noise ratio (SNR) of the communication channel and is a function of the desired bit error ratio (BER). The BER is the number of single bit transmission/reception errors per the total

number of bits transmitted. Increasing the number of discrete amplitudes and/or phases associated with a particular carrier (i.e., increasing the constellation size) increases the likelihood of bit errors. The BER increases with increasing constellation size because, as the number of discrete amplitudes and/or phases increases, the magnitude of the difference between discrete phases and/or amplitudes decreases and hence the ability of the receiver to distinguish between different phase and/or amplitude values decreases.

The relationship between BER and SNR is well-known in the art of multicarrier communication. Tables are available that show the minimum SNR that can support a BER of a fixed amount or less for a given constellation size. For example, the table shown below, $SNR[c_j]$, a constellation signal to noise ratio, indicates the minimum SNR needed to transmit a constellation having the indicated size in order to obtain an expected BER of 10^{-7} (i.e., an error of one bit per every 10^7 bits that are transmitted.) Note that as the constellation size increases, the minimum required SNR also increases.

Constellation size c (in bits)	SNR requirement
2	14 dB
3	19 dB
4	21 dB
5	24 dB

Referring to Fig. 4, a graph 110 illustrates a relationship between SNR and frequency for a communication channel transmitting a multicarrier signal having carriers between frequencies f_1 and f_j. A vertical axis 112 of the graph 110 represents SNR. A horizontal axis 114 of the graph 110 represents frequency in a manner similar to that illustrated in connection with the horizontal axis 102 of the graph 100 of Fig. 3.

A plot 116 shows the relationship between SNR and frequency for the frequencies between f_1 and f_J, the lowest and highest (respectively) carrier frequencies for the multicarrier frequency signal. The plot 116 illustrates that the SNR varies according to frequency so that, for example, the SNR at frequency f_m is lower than the SNR at frequency f_n. Based on the table shown above, it is possible that, for a given BER, the

constellation size supported by the carrier frequency f_m is smaller than the constellation size supported by the carrier frequency f_n.

Referring to Fig. 5, a graph 120 uses a plot 122 to illustrate a hypothetical relationship between SNR and frequency. The graph 120 is similar to the graph 110 of Fig. 4. The vertical axis of the graph 120, which represents SNR, has superimposed thereon the SNR requirement numbers from the table, shown and discussed above, that relates minimum SNR requirements with constellation size for a BER of 10^{-7}. The graph 120 shows that an SNR of 14 dB is required to support a constellation size of two bits and that SNR's of 19, 21, and 24 are required to support constellation sizes of three, four, and five bits, respectively. Based on this, it is possible to use the plot 122 to determine a maximum constellation size for each of the carrier frequencies between f_l and f_j. For example, the plot 122 shows that any carrier frequencies between f_l and f_a can support a maximum constellation size of four bits since all portions of the plot 122 between f_l and f_a are greater than 21 dB (the minimum required SNR to support a constellation size of four bits), but less than 24 dB (the minimum SNR for five bits). No carrier frequencies between f_l and f_a can support a constellation size of five bits at the BER used to generate the minimum SNR requirements.

The portion of the plot 122 between f_a and f_b is shown in Fig. 5 as being greater than 24 dB. Accordingly, carrier frequencies between f_a and f_b can support a maximum constellation size of at least five bits. Similarly, carrier frequencies between f_b and f_c will support a maximum constellation size of four bits; carrier frequencies between f_c and f_d will support a maximum constellation size of three bits; carrier frequencies between f_d and f_e will support a maximum constellation size of two bits, and carrier frequencies between f_e and f_j will support a maximum constellation size of three bits.

The difference between the minimum required SNR and the actual transmission channel SNR is called the "margin". For example, the plot 122 shows that if four bits are used at the carrier frequency f_1, the carrier frequency at f_l will have a margin 124 somewhat greater than zero since the SNR at f_l is shown in Fig. 5 as being greater than

the minimum SNR requirement of 21 dB. Similarly, it is possible to use less than the maximum supported constellation size at a particular carrier frequency. For example, although the plot 122 shows that a carrier at the frequency f_a will support a constellation size of five bits (since the SNR at f_a is 24 dB), it is possible to encode the carrier at the frequency f_a with only three bits. In that case, the margin at the frequency f_a is the difference between the transmission channel SNR at f_a (24 dB) and the SNR required to support a constellation of three bits at frequency f_a (19 dB). Accordingly, the margin at frequency f_a is 5 dB.

In instances where the multicarrier signal is used to transmit the maximum number of data bits, then the SNR of the communication channel is first measured and then each carrier is set to the maximum supported constellation size. However, in many applications, the multicarrier signal is used to transmit less than the maximum possible number of bits. In those cases, it is advantageous to maximize the overall margin of the signal to thus reduce the error rate. This can be illustrated by a simple example:

Assume a two-channel multicarrier signal has a maximum constellation size of five bits for the first carrier and four bits for the second carrier. Further assume that it is desirable to use the signal to transmit six bits. One way to allocate the bits among the two carriers is to use the first carrier to transmit five bits and the second carrier to transmit one bit. In that case, however, the margin for the first carrier is relatively small while the margin for the second carrier is relatively large. There will be many more errors for bits transmitted via the first carrier than bits transmitted via the second carrier and, since most of the bits are being transmitted via the first carrier anyway, then the overall error rate of the signal, while below the target BER, is still higher than it has to be in this case. A more advantageous way to allocate the bits might be to allocate three bits to each of the two carriers. In that case, both of the carriers operate with a relatively large margin and the overall error rate of the signal is reduced.

Of course, in many multicarrier communication applications, there are hundreds of carriers and hundreds to thousands of bits that are transmitted. In addition, it is necessary to allocate the bits in a relatively rapid manner since time spent allocating bits is time not spent communicating information. Furthermore, it may be necessary to reallo-

cate the bits during communication if the channel transmission characteristics change dynamically.

Referring to Fig. 6, a flow chart 150 illustrates a technique for allocating bits among carriers of a multicarrier signal. Processing begins at a first step 152 where various quantities used to allocate the bits are initialized. These quantities include MH, the high-bound for the margin, ML, the low-bound for the margin, and k, an iteration counter which is described in more detail below. Following step 152 is a step 154 where the margin, M, is calculated by averaging M_H and M_L.

Following step 154 is a step 156 where a table indicating required SNR for various constellation sizes, RSNR[c], is calculated. RSNR[c] is a table having entries equal to the sum of the margin, M, and the minimum SNR requirements that can support a constellation of size c, and thus comprises an augmented constellation signal to noise ratio, $SNRa[c_j] = SNR[c_j] + M$. Following step 156 is a step 158 where a bit table, B[j], is calculated. B[j] is a table of the maximum number of bits that can be allocated to each of the carriers $f_1, \ldots f_j$, given the values stored in RSNR[c]. The maximum number of bits are allocated for each carrier in a manner similar to that discussed above in connection with Fig. 5.

Following step 158 is a step 160 where a value N_{max} is calculated. N_{max} represents the maximum number of bits that can be transmitted on the channel and is determined by summing all of the values in the table B[j]. Since the table B[j] contains the maximum number of bits that can be transmitted for each carrier based on the minimum required SNR for each constellation size plus the calculated margin, then N_{max} represents the maximum number of bits that can be transmitted on the channel wherein each of the carriers has a margin of at least M.

Following the step 160 is a test step 162 which determines if N_{max} equals N where N is the number of bits that are to be transmitted using the multicarrier signal. If N_{max} does in fact equal N, then processing is complete and the bit table B[j] represents an allocation of bits among the carriers of the multicarrier signal wherein each carrier will have a margin at least as large as M.

If it is determined at the test step 162 that N_{max} does not equal N, then processing transfers from the test step 162 to a test step 164. Note that if N is less than N_{max}, then the margin can be increased (in order to decrease N_{max}) in the next iteration. Similarly, if N is not less than N_{max}, then the margin is too large and needs to be decreased in the next iteration. If it is determined at the test step 164 that N is less than N_{max}, the control transfers from the test step 164 to a step 166 where M_L, the low-bound on the margin, is set equal to M. Setting M_L equal to M effectively increases M_L, causing an increase in the value of the margin, M, that will be calculated on the next iteration at the step 154.

Conversely, if it is determined at the step 164 that N is not less than N_{max}, then control transfers from the step 164 to a step 168 where M_H, the high-bound on the margin is set equal to M. This effectively decreases the value of M_H, thus causing the value of M to decrease when M is calculated at the step 154 on the next iteration.

Control transfers from either the step 166 or step 168 to a step 170 where the iteration counter, k, is incremented. Following the step 170 is a test step 172 which determines if the iteration counter is less than the maximum allowable value for the iteration counter, K_{max}. The iteration counter, k, is used to ensure that the algorithm will terminate after a certain number of iterations even if the terminating condition at the step 162 (i.e., $N_{max} = N$) is never met. In a preferred embodiment, K_{max} equals 16.

If it is determined at the test step 172 that k is not less than K_{max}, then control transfers from the step 172 to a step 174 where the remaining bits are either removed or added to the bit table, B[j], as appropriate. Bits are added or removed at the step 174 in a random or pseudo random manner so that the sum of all allocated bits in the table B[j], equals N, the number of bits that are to be transmitted via the multichannel signal. Note that in this instance, there is no guarantee that each of the carriers has a margin of at least M. The step 174 is simply executed in order to finalize the allocation process if the algorithm is unable to meet the termination condition at the step 162.

If it is determined at the test step 172 that the iteration counter, k, is less than the predetermined maximum value for the iteration counter, then control transfers from the

step 172 to a test step 176 which determines if the algorithm is diverging, i.e., if (N_{max}-N) is increasing. It is desirable for the algorithm to converge so that the value of N_{max} gets closer to the value of N with each iteration because the algorithm terminates when N_{max} equals N at the test step 162. However, if it is determined at the test step 176 that the value of N_{max} is actually getting farther from the value of N with each iteration, then control transfers from the step 176 to the step 174 where the remaining bits are distributed randomly among the values in the table B[j], as discussed above, after which processing is complete.

If it is determined at the test step 176 that the algorithm is not diverging, then control transfers from the step 176 back to the step 154 where the margin is calculated for the next iteration. The margin calculated at the subsequent iteration 154 will either be less than or greater than the margin calculated on the previous iteration, depending upon whether N was less than N_{max} or not at the test step 164, as discussed above.

Referring to FIG. 7, a flow chart 180 illustrates in detail the initialization routine for the step 152 of the flow chart 150 shown in FIG. 6. The initialization routine is entered and processing begins at a step 182 where the transmission characteristics of the channel are measured to determine the signal-to-noise ratio at each of the carrier frequencies of the multicarrier signal. As discussed above in connection with Fig.'s 4 and 5, the transmission channel signal-to-noise ratio may be a function of frequency. Measuring the channel transmission characteristics at the step 182 is discussed in more detail hereinafter.

Following the step 182 is a step 184 where the minimum required signal-to-noise ratio table, SNR[c], is initialized. As discussed above, for a given bit error ratio (BER), the minimum required SNR for each constellation size, c, can be determined via conventional calculations known in the art or by looking up the values in a textbook. Following the step 184 is a step 186 where the bit table, B[j], is calculated. Calculation of the bit table at the step 186 is similar to calculation of the bit table at the step 158 discussed above in connection with the flow chart 150 of FIG. 6, except that the unaugmented SNR table is $SNR[c_j]$ used at the step 186 rather than the RSNR table which is used at the step 158. Using the SNR table at the step 186 effectively calculates the bit table,

B[j], with a margin of zero. Following the step 186 is a step 188 where N_{max} is calculated. The step 188 is similar to the step 160 discussed above in connection with the flow chart 150 of FIG. 6; N_{max} is simply the sum of all the entries in the bit table, B[j].

Following the step 188 is a step 190 where it is determined if N_{max} equals N. If N_{max} does equal N at the step 190, then processing is complete for the entire algorithm (not just the initialization portion) since the channel will only support N_{max} bits of transmission. That is, if N_{max} equals N at the step 190, there is no point in continuing with the algorithm and calculating a margin since, by default, the channel can transmit no more than N bits.

If it is determined at the test step 190 that N_{max} does not equal N, then control transfers from the step 190 to a test step 192 where it is determined if N is less than N_{max}. Note that if N is not less than N_{max} at the step 192, then the channel will not support transmission of N bits at the BER used to construct the SNR table at the step 184. That is, the bandwidth of the channel is too low. However, in this case, the algorithm can continue by calculating a negative margin and simply proceeding to maximize the negative margin so that, although the BER that will be achieved will exceed the desired BER, it is still minimized given the requested data rate. In another embodiment, the algorithm can terminate at this point and indicate that the bits cannot be allocated. In yet another embodiment, the algorithm can be rerun using a higher BER and (presumable) lower minimum SNR requirements for the various constellation sizes.

If it is determined at the step 192 that N is not less than N_{max} (i.e., the system will be operating with a negative margin) then control transfers from the step 192 to a step 198 where the low-bound on the margin M_L, is set to zero. Following the step 198 is a step 200 where the high-bound on the margin is set using the formula $M_K = (10/J^*)(N_{max}\text{-}N)$. Note that, however, in this case the high-bound on the margin will be set to a positive value at the step 200 because N_{max} - N will be a positive number.

Following either the step 200 or the step 196, control transfers to a step 202 where the iteration counter that is used to terminate the algorithm after a predetermined number of iterations is set to one. Following the step 202, the initialization routine is

exited so that the remaining processing, as discussed above in connection with FIG. 6, can continue.

The formula used to set M_L at the step 196 and to set M_H at the step 200 provides upper and lower bounds of the margin such that the algorithm converges in a reasonable number of iterations while ensuring that the final margin does not fall outside the range between M_L and M_H. Of course, it is possible to practice the invention using other formulas or techniques for calculating initial values for M_L and M_H.

Referring to FIG. 8, a flow chart 210 illustrates operation of software used by the receiver to allocate bits among the different carriers of the multicarrier signal and synchronize changes in the bit allocation table with the transmitter. Processing begins at a first test step 262 which determines if the receiver has received a reference frame. A reference frame is a predetermined and detectable frame of special data bits that is provided by the transmitter to the receiver to allow the receiver to determine the channel characteristics. In a preferred embodiment, the reference frame is transmitted periodically, although other conventional techniques can be used to determine whether the reference frame should be sent by the transmitter. The reference frame is recognized by the receiver using any one of a variety of conventional techniques such as a special header in a packet indicating that a reference frame is being provided. Use of a reference frame in connection with multicarrier communication is well-known in the art. If a reference frame is not received at the step 262, the software loops back to the test step 262 to poll for receipt of the reference frame.

If it is determined at the test step 262 that a reference frame has been received, then control transfers from the step 262 to a step 264 where the errors in the reference frame are measured with respect to the brown constellation distances of the first signal. Note that since the reference frame is a predetermined signal, the receiver can know exactly what was sent by the transmitter. Therefore, any differences between the data received by the receiver and the expected values for signal data can be accounted for by errors induced by the transmission channel. These errors are measured at the step 264.

Following the step 264 is a step 266 where the receiver determines the channel characteristics based on the errors measured at the step 264. This is done in a conventional manner using techniques for determining channel characteristics based on detected

transmission errors. Following the step 266 is a step 268 where the receiver allocates various bits among the carriers using, in a preferred embodiment, the technique disclosed above in connection with Fig.'s 6 and 7.

Following the step 268 is a test step 270 which determines if the bit allocation table provided at the step 268 is different than the previous bit allocation table. That is, it is determined at the step 270 if there is a difference between the recently-calculated bit allocation table and the previous bit allocation table. If it is determined at the test step 270 that there is no difference (i.e., that the bit allocation table has not changed), then control transfers from the step 270 back to the step 262 where the software waits for the transmitter to send another reference frame. Otherwise, if it is determined at the step 270 that the new bit allocation table is different than the old bit allocation table, then control transfers from the step 270 to a step 272 where a flag is sent from the receiver to the transmitter indicating that the bit allocation table has changed. In a preferred embodiment, the flag is sent at the step 272 via a single carrier of the multicarrier signal that is reserved for use by the transmitter and receiver only for the flag. In another embodiment, the reserved carrier can also be used to transmit the new bit allocation table.

Following the step 272 is a step 274 where the receiver sends the new bit allocation table, determined at the step 268, to the transmitter. Following the step 274, control transfers back to the test step 262 to poll and wait for the transmitter to send another reference frame.

While the invention has been disclosed in connection with the preferred embodiments shown and described in detail, various modifications and improvements thereon will become readily apparent tot hose skilled in the art. Accordingly, the spirit and scope of the present invention is to be limited only by the following claims.

CLAIMS

1. In a multicarrier modulation system having a plurality of channels for transmitting data at varying rates from a transmitter to a receiver dependent on the signal to noise ratio of the respective channels, the improvement comprising:

   A. means for allocating data to respective ones of said channels in accordance with an initial signal to noise ratio for the corresponding channel,

   B. means for repetitively calculating trial noise margins across said channels, and

   C. means for repetitively combining said trial noise margins with the said signal to noise ratios of said channels to form modified signal to noise ratios for said channels for use in reallocating said data thereto.

2. A multicarrier modulation system according to claim 1 in which said margins are added to the constellation signal to noise ratios associated with said channels to form said modified ratios.

3. A multicarrier modulation system according to claim 2 in which said margins are added to said signal to noise ratios equally across said channels.

4. A multicarrier modulation system according to claim 3 which includes means for defining upper and lower margin thresholds M_H and M_L, respectively, said trial noise margins being defined as a combination of said thresholds.

5. A multicarrier modulation system according to claim 4 in which said combination is formed as an average of said upper and lower thresholds.

6. A multicarrier modulation system according to claim 5 in which at least one of said thresholds is determined as a function of the difference between the amount of data transmissible across said channels in accordance with previously specified signal to noise ratios associated with said channels and the amount of data desired to be transmitted across said channels.

7. A multicarrier modulation system according to claim 6 in which at least one of said thresholds is set to zero.

8. A multicarrier modulation system according to claim 6 which includes means for terminating data allocation when the amount of data transmissible across said channels in accordance with previously specified signal to noise ratios associated with said channels equals the amount of data desired to be transmitted across said channels.

9. A multicarrier modulation system according to claim 6 which includes means for terminating data allocation when said difference diverges.

10. A multicarrier modulation system according to claim 6 which includes means for terminating data allocation after a defined number of iterations of margin calculations over said channels.

11. A multicarrier modulation system according to claim 1 in which said means for calculating trial noise margins comprises:

A. means for defining a trial margin that is a function of the difference between the amount of data allocable to said channels in accordance with said initial signal to noise ratios for the respective channels and the amount of data desired to be transmitted, and

B. means for repetitively adjusting said trial margin in accordance with the relation between the amount of data transmissible across said channels when the signal to noise ratios of said channels are augmented by said trial margin and the amount of data

C. transmissible across said channels in accordance with a prior determination of said signal to noise ratios.

12. A multicarrier modulation system according to claim 1 which includes means for periodically transmitting a reference frame from the transmitter to the receiver across said channels, and means for measuring the signal to noise ratios of said channels from the transmitted reference frame, said means for repetitively calculating trial noise margins across said channels using the signal to noise ratios determined in the most recently transmitted frame as the initial signal to noise ratios for calculating said margins in the interval between said frame and the next frame.

13. A multicarrier modulation system according to claim 12 which includes first and second memory register sets at both said transmitter and said receiver for storing channel data allocations in accordance with signal to noise ratios associated therewith, and means for transmitting from the transmitter to the receiver a flag indicating which of the register sets is to be used for subsequently receiving data from said transmitter.

14. In a multicarrier modulation system having a plurality of channels for transmitting data at varying rates from a transmitter to a receiver dependent on the signal to noise ratio of the respective channels, the improvement comprising:

A. means for allocating data to respective ones of said channels in accordance with initial signal to noise ratios measured for the corresponding channels,

B. means for calculating a trial noise margin across said channels as a function of said initial signal to noise ratio and the difference between the amount of data transmissible over said channels with said signal to noise ratios and the amount of data desired to be transmitted,

C. means for augmenting the initial signal to noise ratios associated with said channel by the trial noise margin to thereby define an augmented signal to noise ratio for use in defining a revised estimate of the amount of data transmissible over said channels,

D. means for repetitively defining successive trial noise margins as a function of the augmented signal to noise ratios and the difference between the

amount of data transmissible over said channels with said augmented signal to noise ratios and the amount of data desired to be transmitted until an exit condition is reached.

15. A multicarrier modulation system according to claim 14 in which said exit condition comprises equality between the amount of data transmissible over said channels with a particular set of augmented signal to noise ratios and the amount of data desired to be transmitted.

16. A multicarrier modulation system according to claim 14 in which said exit condition comprises an increase in the difference between the amount of data transmissible over said channels with said augmented signal to noise ratios and the amount of data desired to be transmitted as determined on successive calculations.

17. A multicarrier modulation system according to claim 14 in which said exit condition comprises determination of a defined number of successive trial noise margins.

18. A multicarrier modulation system according to claim 14 which includes means for periodically transmitting a reference frame from the transmitter to the receiver across said channels, and means for measuring the signal to noise ratios of said channels from the transmitted reference frame, said means for calculating trial noise margins across said channels using the signal to noise ratios determined in the most recently transmitted frame as the initial signal to noise ratios for calculating said margins in the interval between said frame and the next frame.

19. A multicarrier modulation system according to claim 18 which includes first and second memory register sets at both said transmitter and said receiver for storing channel data allocations in accordance with signal to noise ratios associated therewith, and means for transmitting from the transmitter to the receiver a flag indicating which of the register sets is to be used for subsequently receiving data from said transmitter.

20. A method of allocating data to respective ones of channels in a multicarrier modulation system having a plurality of channels for transmitting data at varying rates from a transmitter to a receiver, comprising the steps of :

A. allocating data to respective ones of said channels in accordance with measured signal to noise ratio for the corresponding channel,

B. repetitively calculating trial noise margins across said channels, and

C. repetitively combining said trial noise margins with the said signal to noise ratios of said channels to form modified signal to noise ratios for said channels for use in reallocating said data thereto.

21. A method according to claim 20 in which the step of combining said trial noise margins and said signal to noise ratios of said channels comprises adding a calculated trial noise margin to the constellation signal to noise ratios of said channels to thereby form an augmented signal to noise ratio from which the amount of data transmissible in said channel is determined.

22. A method according to claim 20 in which the step of repetitively calculating trial noise margins across said channels comprises the steps of

A. repetitively defining a trial margin that is a function of the difference between the amount of data allocable to said channels in accordance with said initial signal to noise ratios for the respective channels and the amount of data desired to be transmitted, and

B. repetitively adjusting said trial margin in accordance with the relation between the amount of data transmissible across said channels when the signal to noise ratios of said channels are augmented by said trial margin and the amount of data transmissible across said channels in accordance with a prior determination of said signal to noise ratios.

23. A method according to claim 22 which further includes the steps of:

A. periodically transmitting a reference frame from the transmitter to the receiver across said channels,

B. measuring the signal to noise ratios of said channels from the transmitted reference frame and using the signal to noise ratios determined in the most recently transmitted frame as the signal to noise ratios for calculating said margins in the interval between said frame and the next frame.

24. A method according to claim 23 which includes the steps of providing first and second memory register sets at both said transmitter and said receiver for storing channel data allocations in accordance with signal to noise ratios associated therewith, and transmitting from the transmitter to the receiver a flag indicating which of the register sets is to be used for subsequently receiving data from said transmitter.

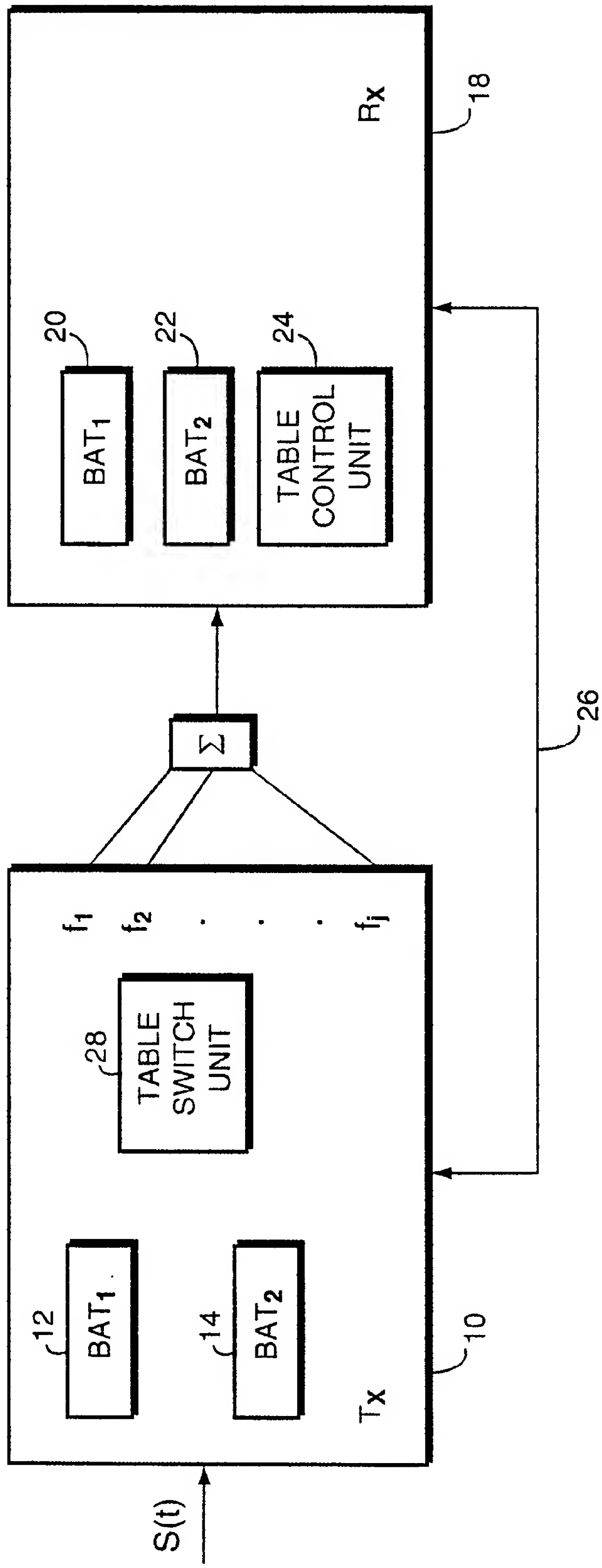
Rx
18
20
BAT1
22
BAT2
24
TABLE CONTROL UNIT
Σ
26
f1
f2
fj
28
TABLE SWITCH UNIT
12
BAT1
14
BAT2
Tx
10
S(t)

FIG. 1

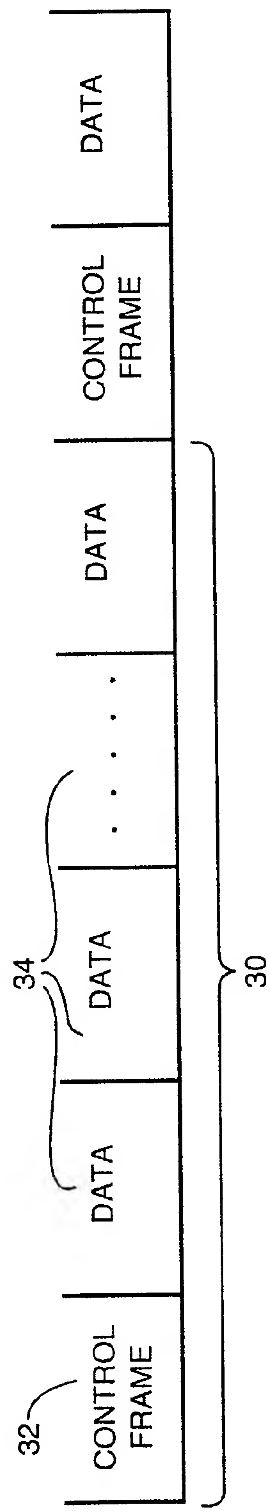

FIG. 2

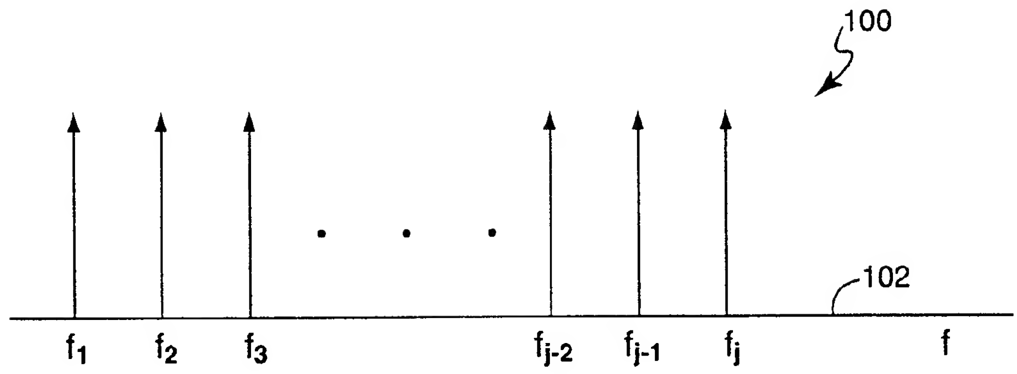

FIG. 3

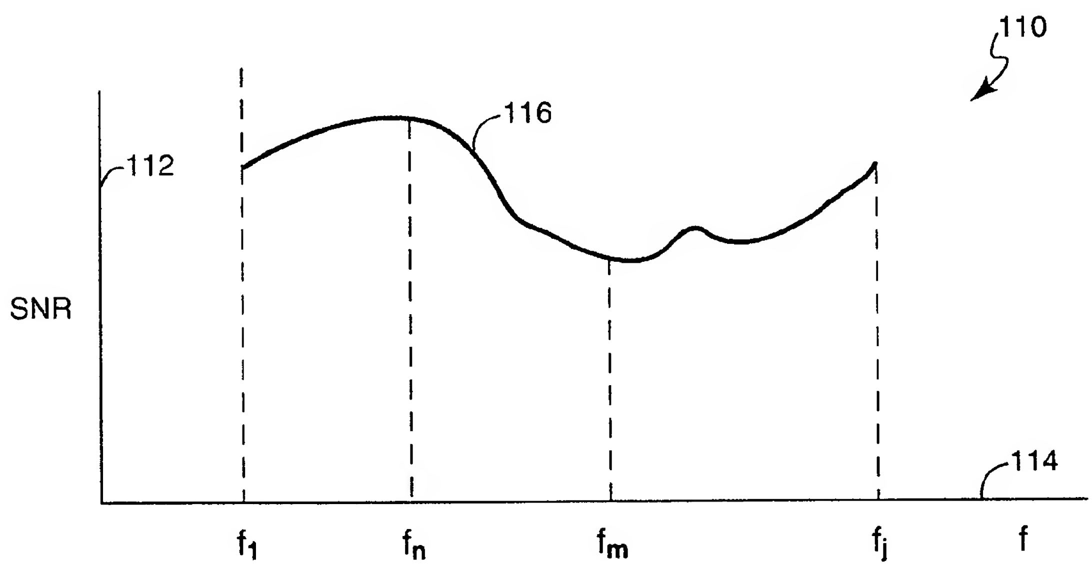

FIG. 4

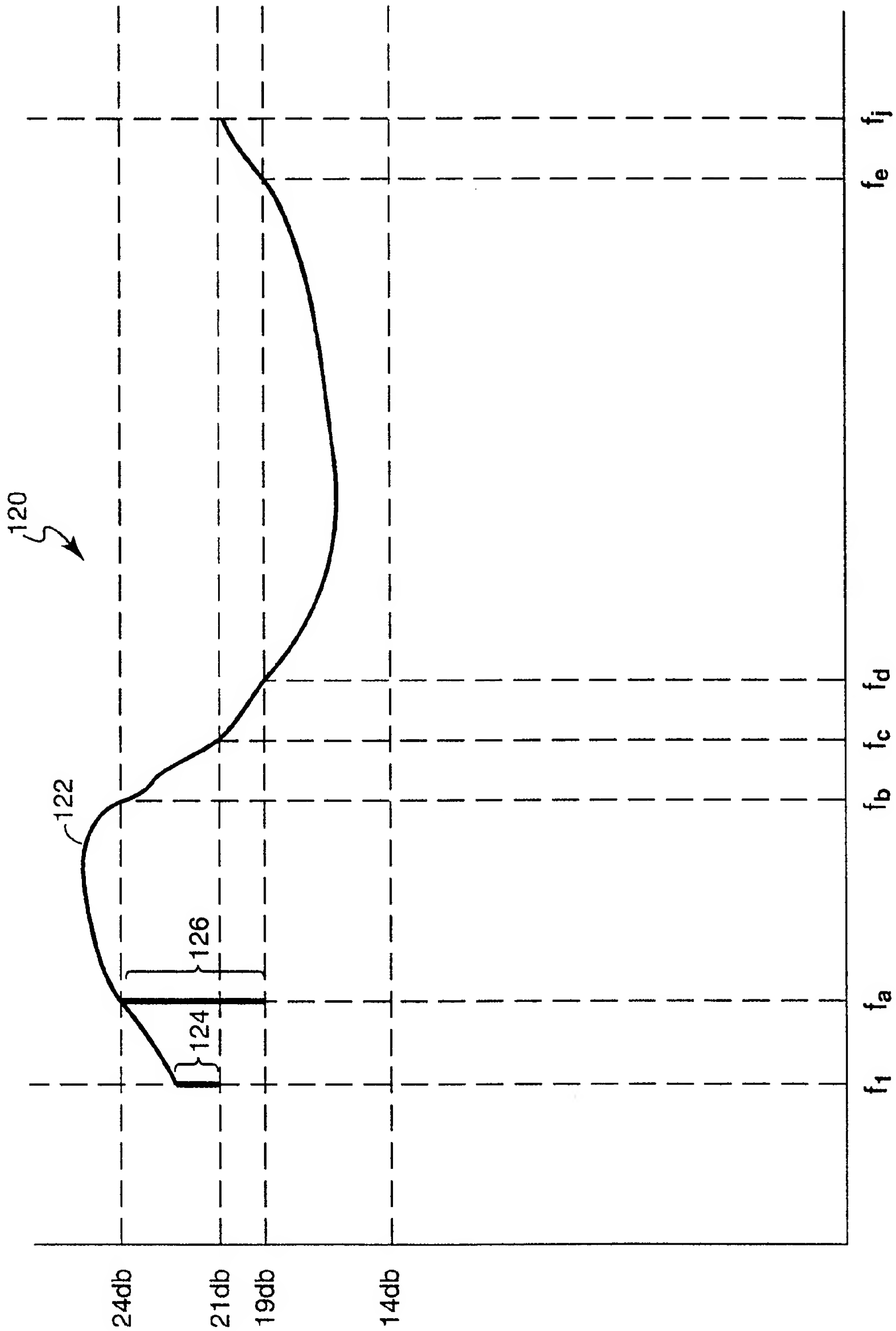

FIG. 5

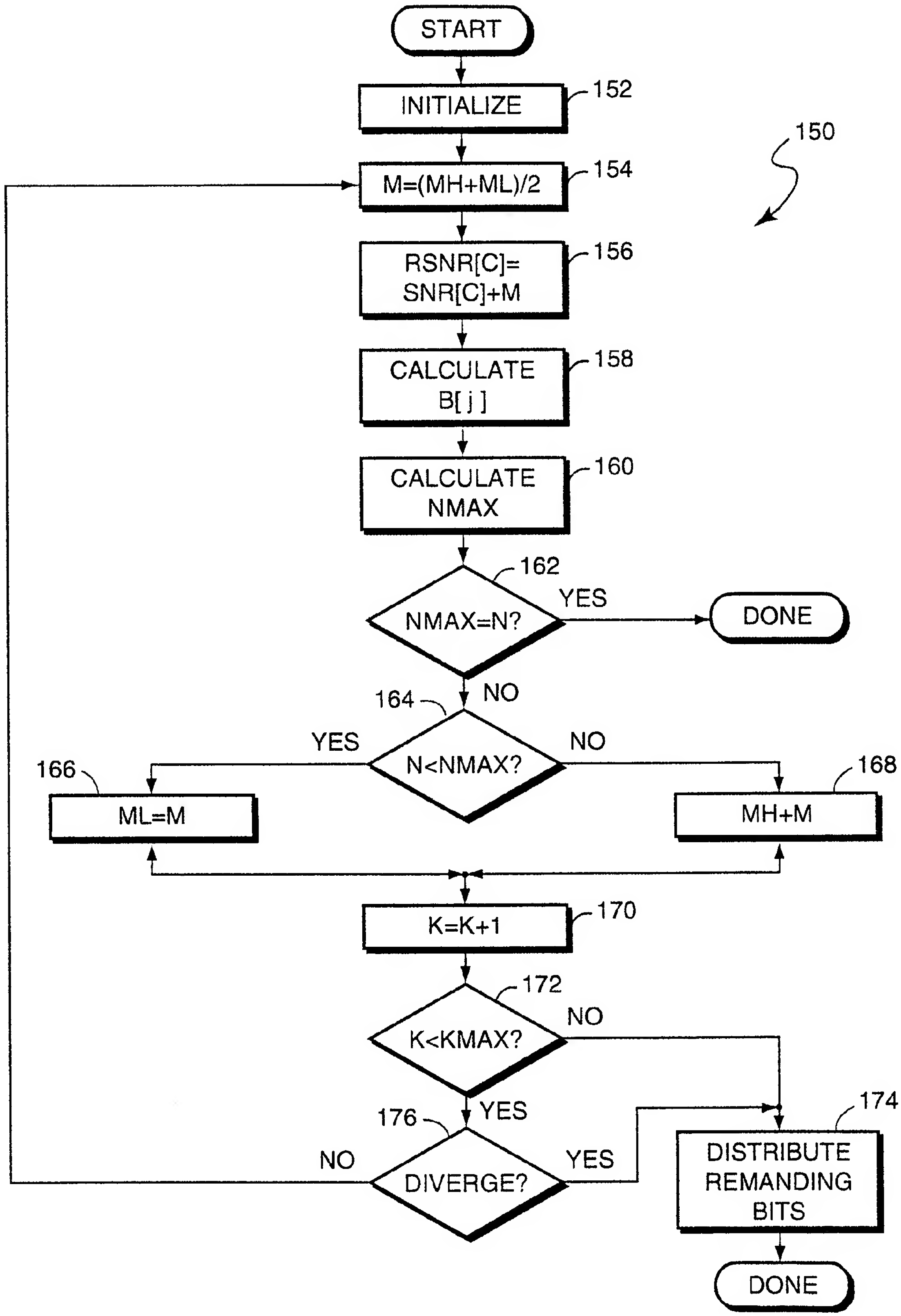
START
INITIALIZE
152
150
M=(MH+ML)/2
154
RSNR[C]=
SNR[C]+M
156
CALCULATE
B[j]
158
CALCULATE
NMAX
160
162
NMAX=N?
YES
DONE
NO
164
N<NMAX?
YES
NO
166
ML=M
168
MH+M
K=K+1
170
172
K<KMAX?
NO
YES
176
174
DIVERGE?
NO
YES
DISTRIBUTE
REMANDING
BITS
DONE

FIG. 6

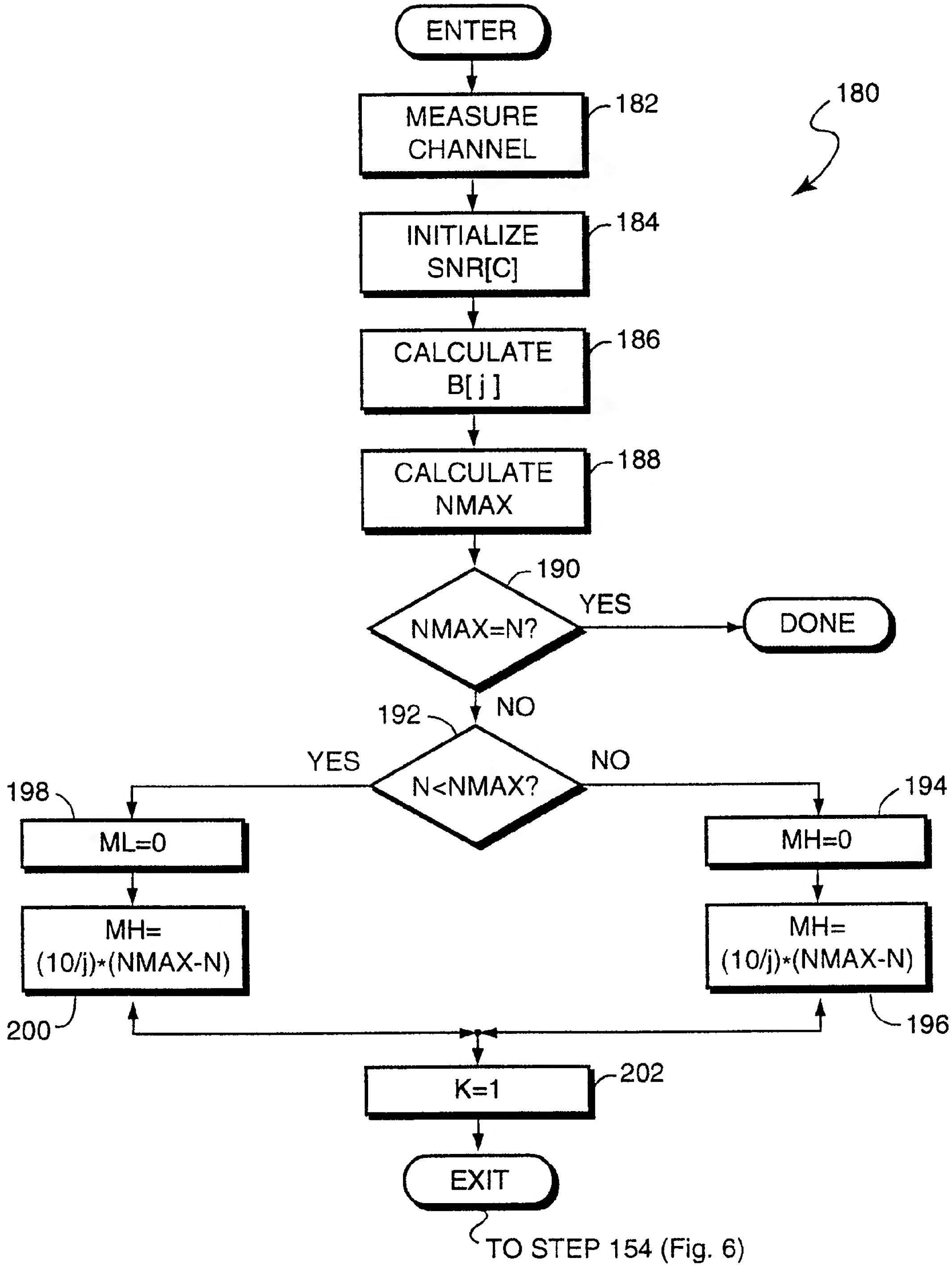
ENTER
MEASURE CHANNEL
182
180
INITIALIZE SNR[C]
184
CALCULATE B[j]
186
CALCULATE NMAX
188
190
NMAX=N?
YES
DONE
NO
192
N<NMAX?
YES
NO
198
ML=0
MH= (10/j)*(NMAX-N)
200
194
MH=0
MH= (10/j)*(NMAX-N)
196
K=1
202
EXIT
TO STEP 154 (Fig. 6)

FIG. 7

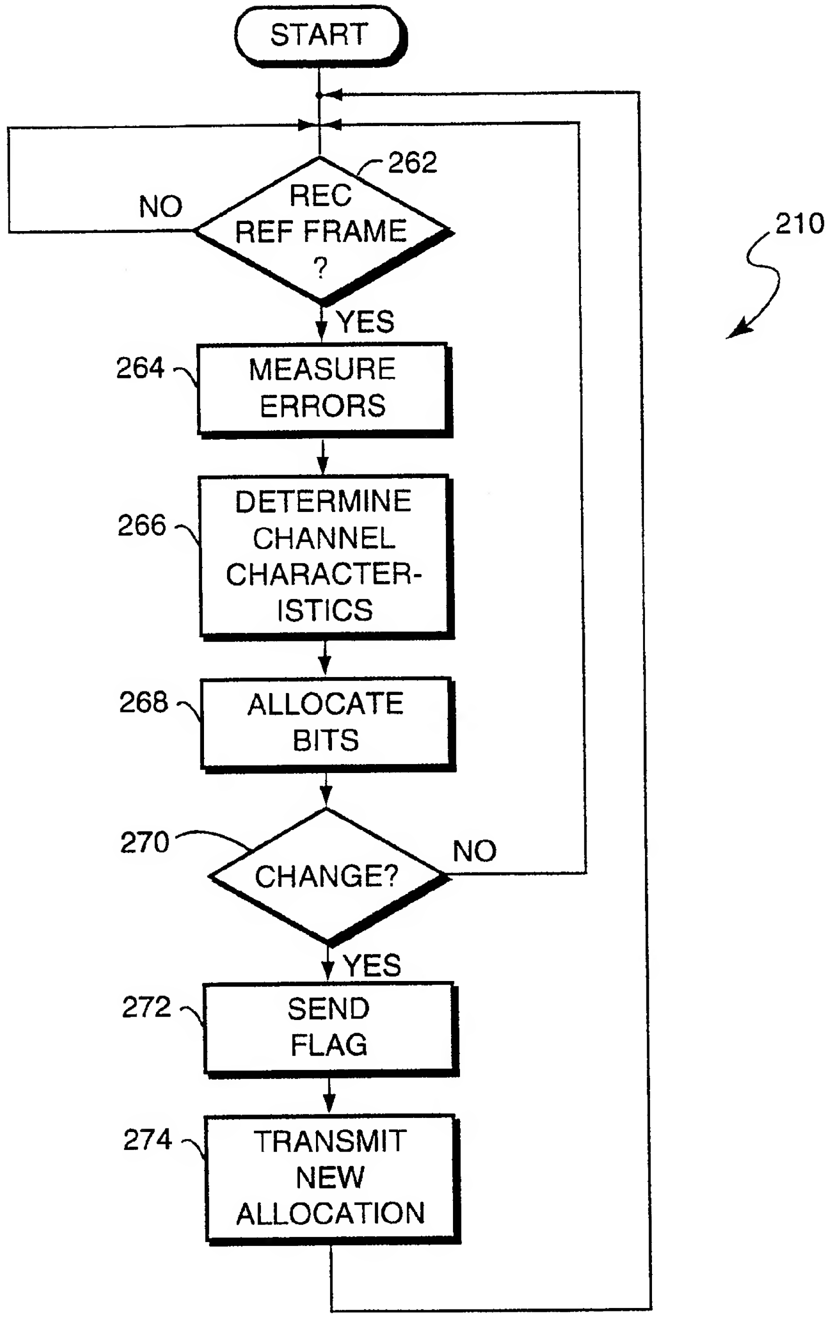
START
262
REC REF FRAME ?
NO
YES
210
264
MEASURE ERRORS
266
DETERMINE CHANNEL CHARACTER-ISTICS
268
ALLOCATE BITS
270
CHANGE?
NO
YES
272
SEND FLAG
274
TRANSMIT NEW ALLOCATION

FIG. 8

INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 98/11845

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 H04L27/26

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	EP 0 753 947 A (ALCATEL BELL NV) 15 January 1997 see column 6, line 39 - line 51 see column 11, line 49 - column 13, line 19 see claims 1-6,8 see figure 3	1-24
A	US 5 479 447 A (CIOFFI JOHN M ET AL) 26 December 1995 cited in the application see column 3, line 57 - column 4, line 28 see column 5, line 42 - column 6, line 14	1-24
A	WO 86 07223 A (TELEBIT CORP) 4 December 1986 cited in the application see page 17, line 12 - page 24, line 2	1-24

-/--

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
12 November 1998	19/11/1998

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Koukourlis, S

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INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 98/11845

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
P,X	EP 0 812 087 A (MOTOROLA INC) 10 December 1997 see the whole document -----	1-24

1

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/US 98/11845

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
EP 0753947 A	15-01-1997	AU 5832696 A	23-01-1997
		CA 2180973 A	12-01-1997
		US 5812599 A	22-09-1998
US 5479447 A	26-12-1995	NONE	
WO 8607223 A	04-12-1986	US 4679227 A	07-07-1987
		AU 3528189 A	21-09-1989
		AU 587037 B	03-08-1989
		AU 5817786 A	24-12-1986
		AU 609355 B	26-04-1991
		AU 6009690 A	15-11-1990
		BR 8606677 A	11-08-1987
		CA 1251586 A	21-03-1989
		CN 1011461 B	30-01-1991
		CN 1048774 A	23-01-1991
		CN 1048775 A,B	23-01-1991
		CN 1048776 A,B	23-01-1991
		DE 3681887 A	14-11-1991
		DK 31087 A	20-01-1987
		EP 0224556 A	10-06-1987
		JP 6003956 B	12-01-1994
		JP 62502932 T	19-11-1987
		PT 82600 B	29-07-1994
		US 4731816 A	15-03-1988
		US 4833706 A	23-05-1989
		US 5054034 A	01-10-1991
EP 0812087 A	10-12-1997	JP 10155031 A	09-06-1998

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A1

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Published
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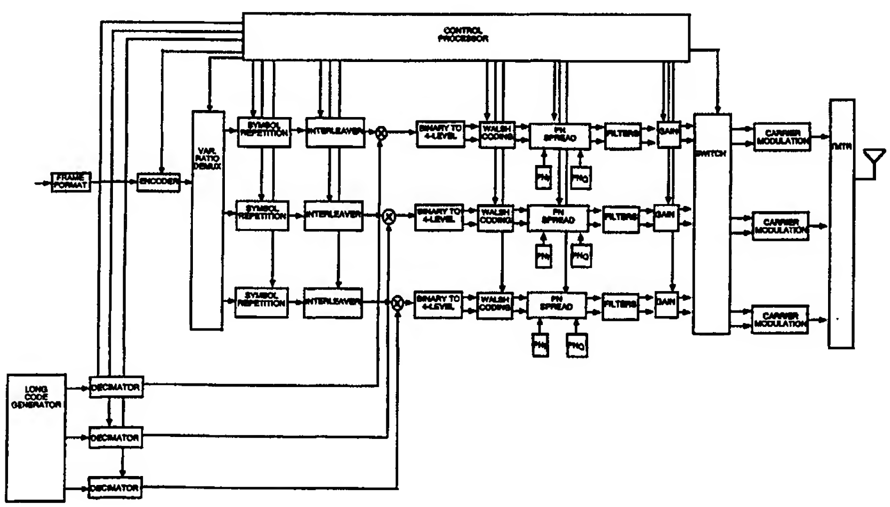
(54) Title: A METHOD OF AND APPARATUS FOR TRANSMITTING DATA IN A MULTIPLE CARRIER SYSTEM



(57) Abstract

A method of an apparatus for transmitting data in a multiple carrier system comprises encoding data and dividing the resulting encoded symbols for transmission on different frequencies. The transmitter comprises a control processor (50) for determining the capacity of each of a plurality of channels and selecting a data rate for each channel depending on the determined capacity. A plurality of transmission subsystems (56 to 72) are responsive to the control processor (50). Each transmission subsystem is associated with a respective one of the plurality of channels for scrambling encoded data with codes unique to the channel for transmission in the channel. A variable demultiplexer (56) under the control of the control processor (50) demultiplexes the encoded data into the plurality of transmission subsystems at a demultiplexing rate derived from the data rates selected for the channels by the controller. In one embodiment of the transmission subsystems, the encoded symbols are provided to a symbol repetition unit (58) which keeps the symbol rate of data to be transmitted fixed. In another embodiment, no symbol repetition is provided and variable length Walsh sequences are used to handle data rate variations.

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A METHOD OF AND APPARATUS FOR TRANSMITTING DATA IN A MULTIPLE CARRIER SYSTEM

BACKGROUND OF THE INVENTION

I. Field of the Invention

The present invention relates to a method of and apparatus for transmitting data in a multiple carrier system. The present invention may be used for maximizing system throughput and increasing signal diversity by dynamically multiplexing signals onto multiple carriers in a spread spectrum communication system.

II. Description of the Related Art

It is desirable to be able to transmit data at rates which are higher than the maximum data rate of a single CDMA channel. A traditional CDMA channel (as standardized for cellular communication in the United States) is capable of carry digital data at a maximum rate of 9.6 bits per second using a 64 bit Walsh spreading function at 1.2288 MHz.

Many solutions to this problem have been proposed. One solution is to allocate multiple channels to the users and allow those users to transmit and receive data in parallel on the plurality of channels available to them. Two methods for providing multiple CDMA channels for use by a single user are described in co-pending U.S. Patent Application Serial No. 08/431,180, entitled "METHOD AND APPARATUS FOR PROVIDING VARIABLE RATE DATA IN A COMMUNICATIONS SYSTEM USING STATISTICAL MULTIPLEXING", filed April 28, 1997 and U.S. Patent Application Serial No. 08/838,240, entitled "METHOD AND APPARATUS FOR PROVIDING VARIABLE RATE DATA IN A COMMUNICATIONS SYSTEM USING NON-ORTHOGONAL OVERFLOW CHANNELS", filed April 16, 1997, both of which are assigned to the assignee of the present invention and are incorporated by reference herein. In addition, frequency diversity can be obtained by transmitting data over multiple spread spectrum channels that are separated from one another in frequency. A method and apparatus for redundantly transmitting data over multiple CDMA channels is described in U.S. Patent No. 5,166,951, entitled "HIGH CAPACITY SPREAD SPECTRUM CHANNEL", which is incorporated by reference herein.

The use of code division multiple access (CDMA) modulation techniques is one of several techniques for facilitating communications in which a large number of system users are present. Other multiple access communication system techniques, such as time division multiple access (TDMA), frequency division multiple access (FDMA) and AM modulation schemes such as amplitude companded single sideband (ACSSB) are known in the art. However, the spread spectrum modulation technique of CDMA has significant advantages over these other modulation techniques for multiple access communication systems.

The use of CDMA techniques in a multiple access communication system is disclosed in U.S. Patent No. 4,901,307, entitled "SPREAD SPECTRUM MULTIPLE ACCESS COMMUNICATION SYSTEM USING SATELLITE OR TERRESTRIAL REPEATERS", assigned to the assignee of the present invention and incorporated by reference herein. The use of CDMA techniques in a multiple access communication system is further disclosed in U.S. Patent No. 5,103,459, entitled "SYSTEM AND METHOD FOR GENERATING SIGNAL WAVEFORMS IN A CDMA CELLULAR TELEPHONE SYSTEM", assigned to the assignee of the present invention and incorporated by reference herein. Code division multiple access communications systems have been standardized in the United States in Telecommunications Industry Association Interim Standard IS-95, entitled "Mobile Station-Base Station Compatibility Standard for Dual Mode Wideband Spread Spectrum Cellular System", which is incorporated by reference herein.

The CDMA waveform by its inherent nature of being a wideband signal offers a form of frequency diversity by spreading the signal energy over a wide bandwidth. Therefore, frequency selective fading affects only a small part of the CDMA signal bandwidth. Space or path diversity on the forward/reverse link is obtained by providing multiple signal paths through simultaneous links to/from a mobile user through two or more antennas, cell sectors or cell-sites. Furthermore, path diversity may be obtained by exploiting the multipath environment through spread spectrum processing by allowing a signal arriving with different propagation delays to be received and processed separately. Examples of the utilization of path diversity are illustrated in co-pending U.S. Patent No. 5,101,501 entitled "SOFT HANDOFF IN A CDMA CELLULAR TELEPHONE SYSTEM", and U.S. Patent No. 5,109,390 entitled "DIVERSITY RECEIVER IN A CDMA CELLULAR TELEPHONE SYSTEM", both assigned to the assignee of the present invention and incorporated by reference herein.

FIG. 1 illustrates a transmission scheme for a multiple-carrier code division multiple access (CDMA) system in which each carrier carries a fixed fraction of the transmitted data. Variable rate frame of information bits are provided to encoder **2** which encodes the bits in accordance with a convolutional encoding format. The encoded symbols are provided to symbol repetition means **4**. Symbol repetition means **4** repeats the encoded symbols so as to provide a fixed rate of symbols out of symbol repetition means **4**, regardless of the rate of the information bits.

The repeated symbols are provided to block interleaver **6** which rearranges the sequence in which the symbols are to be transmitted. The interleaving process, coupled with the forward error correction, provides time diversity which aids in the reception and error recovery of the transmitted signal in the face of burst errors. The interleaved symbols are provided to data scrambler **12**. Data scrambler **12** multiplies each interleaved symbol by +1 or -1 according to a pseudonoise (PN) sequence. The pseudonoise sequence is provided by passing a long PN sequence generated by long code generator **8** at the chip rate through decimator **10** which selectively provides a subset of the chips of the long code sequence at the rate of the interleaved symbol stream.

The data from data scrambler **12** is provided to demultiplexer (DEMUX) **14**. Demultiplexer **14** divides the data stream into three equal sub-streams. The first sub-stream is provided to transmission subsystem **15a**, the second sub-stream to transmission subsystem **15b** and the third sub-stream to transmission subsystem **15c**. The subframes are provided to serial-to-parallel converters (BINARY TO 4 LEVEL) **16a-16c**. The outputs of serial to parallel converters **16a-16c** are quaternary symbols (2bits/symbol) to be transmitted in a QPSK modulation format

The signals from serial-to-parallel converters **16a-16c** are provided to Walsh coders **18a-18c**. In Walsh coders **18a-18c**, the signal from each converter **16a-16k** is multiplied by a Walsh sequence consisting of ±1 values. The Walsh coded data is provided to QPSK spreaders **20a-20c**, which spread the data in accordance with two short PN sequences. The short PN sequence spread signals are provided to amplifiers **22a-22b** which amplify the signals in accordance with a gain factor.

The system described above suffers from a plurality of drawbacks. First, because the data is to be provided in equal sub-streams on each of the carriers, the available numerology is limited to frames with a number of code symbols that will divide evenly by a factor of three. Table 1 below

illustrates the limited number of possible rate sets which are available using the transmission system illustrated in FIG. 1.

Walsh Function (QPSK Symbol) Rate [sps]	Number of Walsh Functions per 20ms		Length of Walsh Function [chips]	Symbol Rate [sps] (After Repetition)	Number of Symbols per 20 ms.	
1228800	24576	$3*(2^{13})$	1	2457600	49152	$3*(2^{14})$
614400	12288	$3*(2^{12})$	2	1228800	24576	$3*(2^{13})$
307200	6144	$3*(2^{11})$	4	614400	12288	$3*(2^{12})$
153600	3072	$3*(2^{10})$	8	307200	61444	$3*(2^{11})$
76800	1536	$3*(2^{9})$	16	153600	3072	$3*(2^{10})$
38400	768	$3*(2^{8})$	32	76800	1536	$3*(2^{9})$
19200	384	$3*(2^{7})$	64	38400	768	$3*(2^{8})$
9600	192	$3*(2^{6})$	128	19200	384	$3*(2^{7})$
4800	96	$3*(2^{5})$	256	9600	192	$3*(2^{6})$
2400	48	$3*(2^{4})$	512	4800	96	$3*(2^{5})$
1200	24	$3*(2^{3})$	1024	2400	48	$3*(2^{4})$
600	12	$3*(2^{2})$	2048	1200	24	$3*(2^{3})$
300	6	$3*(2^{1})$	4096	600	12	$3*(2^{2})$
150	3	$3*(2^{0})$	8192	300	6	$3*(2^{1})$

Table 1

As illustrated in Table 1, because the symbols are evenly distributed to the three carriers, the total data rate is limited by the carrier with the least power available or requiring the highest SNR. That is the total data rate is equalt to three times te data rate of the "worst" link (here the worst means the one requiring the highest SNR or having the least power avaialble). this reduces the system throughput, because the worst link's rate is always chosen as the common rate for all three carriers, which results in under utilization of the channel resource on the two better links.

Second, frequency dependent fading can severely affect one of the frequencies while having a limited effect on the remaining frequencies. This implementation is inflexible and does not allow transmission of a frame to be provided in a way that reduces the effects of the poor channel. Third, because of frequency dependent fading, the fading will typically always affectthe same groups of symbolsof each frame. Fourth, were the

implementation to be superimposed on a speech transmission system there is no good way to balance the loads carried on the different frequencies on a frame by frame basis in the face of variable speech activities in each frame. This results in loss in total system throughput. And fifth, for a system with only three frequency channels, with the implementation described, there is no method of separating the speech and data so as to provide the data on one frequency or set of frequencies and the speech on a different frequency or set of frequencies. This results in a loss of system throughput as mentioned above.

Therefore, there is a need felt for an improved multi-carrier CDMA communication system which offers greater flexibility in numerology and load balancing, better resolution in data rates supported, and which offers superior performance in the face of frequency dependent fading and uneven loading.

SUMMARY OF THE INVENTION

In one aspect the invention provides a transmitter for transmitting data at a data rate in a plurality of channels each having a capacity less than the data rate, the transmitter comprising: a controller for determining the capacity of each of a plurality channels and selecting a data rate for each channel depending on the determined capacity; a plurality of transmission subsystems responsive to the controller and each associated with a respective one of the plurality of channels for scrambling encoded data with codes unique to the channel for transmission in the channel; and a variable demultilplexer responsive to the controller for demultiplexing the encoded data into the plurality of transmission subsystems at a demultiplexing rate derived from the data rates selected for the channels by the controller.

In another aspect the invention provides a receiver comprising: a receiving circuit for receiving signals simultaneously in a plurality of channels each of which signals define scrambled encoded symbols which together represent data from a common origin; a controller for determining a symbol rate for the signals in each channel; a plurality of receiving subsystems responsive to the controller and each associated with a respective one of the plurality of channels for descrambling encoded symbols with codes unique to the channel to enable the data to be extracted therefrom; and a variable multilplexer responsive to the controller for multiplexing the data from the plurality of receiving subsystems at a

multiplexing rate derived from the symbol rates determined for the channels by the controller onto an output.

In a further aspect the invention provides a wireless transmitter, comprising: encoder for receive a set of information bits and encoding said information bits to provide a set of code symbols; and a transmission subsystem for receiving said code symbols and for providing a subset of said code symbols on a first carrier frequency and the remaining symbols on at least one additional carrier frequency.

The invention also provides a method of transmitting data at a data rate in a plurality of channels each having a capacity less than the data rate, the method comprising: determining the capacity of each of a plurality channels and selecting a data rate for each channel depending on the determined capacity; scrambling encoded data with codes unique to the channel for transmission in the channel; and demultiplexing the encoded data into the plurality of channels at a demultiplexing rate derived from the data rates selected for the channels by the controller.

The invention further provides a method of receiving data, the method comprising: receiving signals simultaneously in a plurality of channels each of which signals define scrambled encoded symbols which together represent data from a common origin; determining a symbol rate for the signals in each channel; descrambling encoded symbols in each channel with codes unique to the channel to enable the data to be extracted therefrom; and multiplexing the descrambled data from the plurality of channels at a multiplexing rate derived from the symbol rates determined for the channels.

To better utilize the channel resource, it's necessary to be able to transmit a different data rate on each carrier according to the channel condition and the available power on each channel. One way of doing this is by changing the ratio of the inverse-multiplexing on to each of the carriers. Instead of distributing the symbols with a ratio of 1:1:1, a more arbitrary ratio can be used together with different repetition schemes as long as the resulted symbol rate on each carrier is a factor of some Walsh function rate. Walsh function rate can be 1228800, 614400, 307200,..., 75 for Walsh function length from 1 to 16384.

Given the Walsh function length, if the symbol rate is lower than the Walsh function rate, symbol repetition is used to "match" the rate. The repetition factor can be any number, integer or fractional. It will be understood by one skilled in the art that when repetition is present, the total transmit power can be proportionately reduced to keep the code symbol

energy constant. The Walsh function length may or may not be the same on the three carriers, depending on whether we need to save code channels. For example, if the supportable code symbol rate on the three channels are 153600 sps, 30720 sps and 102400 sps (for rate 1/2 coding, these correspond to data rates of 76.8 kbps, 15.36 kbps and 51.2 kbps, respectively - the total data rate is 143.36 kbps), then the inverse-multiplexing ratio will be 15:3:10.

If a Walsh function of length 8 is used for all three channels (assuming QPSK modulation with a QPSK symbol rate of 153.6 Ksps), then each code symbol is transmitted twice, 10 times, and three times on the three channels, respectively. Additional time diversity can be obtained if the repeated symbols are further interleaved. In an alternative embodiment, different Walsh function lengths are used. For example, Walsh functions for the three channels in the example of above of length 16, 16 and 8 respectively can be used, with each code symbol transmitted once on the first channel, five times on the second, and three times on the third.

The above approach does not affect the encoder since it has to be able to handle the highest data rate anyway. All that is changed is the number of data octets at the encoder input. However, this approach does have an impact on the implementation of the interleaver because the interleaver will have many possible sizes (in terms of number of symbols) if all combinations of data rates on the three channels are allowed. One alternative to the above approach which mitigates this problem is to inverse-multiplex the code symbols out of the encoder to the three carriers directly and perform interleaving of repeated code symbols on each channel separately. This simplifies the numerology and reduces the number of possible interleaver sizes on each channel.

BRIEF DESCRIPTION OF THE DRAWINGS

Further features, objects, and advantages of the present invention will become more apparent from the detailed description set forth below of embodiments of the invention when taken in conjunction with the drawings in which like reference characters identify correspondingly throughout and wherein:

FIG. 1 is a block diagram illustrating a multiple frequency CDMA communication system with fixed rates and carriers;

FIG. 2 is a block diagram illustrating a transmission system embodying the present invention;

FIG. 3 is a block diagram illustrating a receiver system embodying the present invention; and

FIG. 4 is a table of code channel Walsh symbols in a traditional IS-95 CDMA communication system.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

Referring to FIG. 2, which is a block diagram illustrating a transmission system embodying the present invention, the first operation to be performed is to determine the amount of data which can be supported on each of the carriers. Three such carriers are illustrated in FIG. 2, though one skilled in the art will realize that the present invention is easily extended to any number of carriers. Control processor **50** based on a set of factors such as the loading on each of the carriers, the amount of data queued for transmission to the mobile station and the priority of the information to be transmitted to the mobile station determines the rate of data transmission on each of the carriers.

After having selected the data rate to be transmitted on each of the carriers, control processor **50** selects a modulation format that is capable of transmitting data at the selected rate. In the exemplary embodiment, different length Walsh sequences are used to modulate the data depending on the rate of the data to be transmitted. The use of different length Walsh sequences selected to modulate the data depending on the rate of the data to be transmitted is described in detail in co-pending U.S. Patent Application Serial No. 08/654,443, filed May 28, 1996, entitled "HIGH RATE DATA WIRELESS COMMUNICATION SYSTEM", which is assigned to the assignee of the present invention and incorporated by reference herein. In an alternative embodiment, the high rate data can be supported by bundling of CDMA channels as described in the aforementioned Patent Applications Serial Nos. 08/431,180 and 08/838,240.

Once the rates which will be supported on each of the carriers is selected then control processor **50** calculates an inverse multiplexing ratio that will determine the amount of each transmission that will be carried on each of the carriers. For example, if the supportable code symbol rate on the three channels are 153600 sps, 30720 sps and 102400 sps (for rate 1/2 coding, these correspond to data rates of 76.8 kbps, 15.36 kbps and 51.2 kbps, respectively - the total data rate is 143.36 kbps), then the inverse-multiplexing ratio will be 15:3:10.

In the exemplary embodiment, frames of information bits are provided to frame formatter **52**. In the exemplary embodiment, formatter **52** generates and appends to the frame a set of cyclic redundancy check (CRC) bits. In addition, formatter **52** appends a predetermined set of tail bits. The implementation and design of frame formatters are well known in the art, an example of a typical frame formatter is described in detail in U.S. Patent No. 5,600,754, entitled "METHOD AND SYSTEM FOR THE ARRANGEMENT OF VOCODER DATA FOR THE MASKING OF TRANSMISSION CHANNEL INDUCED ERRORS", which is assigned to the assignee of the present invention and incorporated by reference herein.

The formatted data is provided to encoder **54**. In the exemplary embodiment, encoder **54** is a convolutional encoder, though the present invention can be extended to other forms of encoding. A signal from control processor **50** indicates to encoder **54** the number of bits to be encoded in this transmission cycle. In the exemplary embodiment, encoder **54** is a rate 1/4 convolutional encoder with a constraint length of 9. It should be noted that because of the additional flexibility provided by the present invention, essentially any encoding format can be used.

The encoded symbols from encoder **54** are provided to variable ratio de-multiplexer **56**. Variable ratio de-multiplexer **56** provides the encoded symbols to a set of outputs based on a symbol output signal provided by control processor **50**. In the exemplary embodiment, there are three carrier frequencies and control processor **50** provides a signal indicative of the number of encoded symbols to be provided on each of the three outputs. As one skilled in the art will appreciate, the present invention is easily extended to an arbitrary number of frequencies.

The encoded symbols provided on each of the outputs of variable ratio de-multiplexer **56** are provided to a corresponding symbol repetition means **58a-58c**. Symbol repetition means **58a-58c** generate repeated versions of the encoded symbols so that the resultant symbol rate matches with the rate of data supported on that carrier and the in particular matches Walsh function rate used on that carrier. The implementation of repetition generators **58a-58c** is known in the art and an example of such is described in detail in U.S. Patent No. 5,629,955, entitled "Variable Response Filter", which is assigned to the assignee of the present invention and incorporated by reference herein. Control processor **50** provides a separate signal to each repetition generator **58a-58c** indicating the rate of symbols on each carrier or alternatively the amount of repetition to be provided on each carrier. In response to the signal from control processor **50**, repetition means **58a-58c**

generate the requisite numbers of repeated symbols to provide the designated symbol rates. It should be noted that in the preferred embodiment, the amount of repetition is not limited to integer number wherein all symbols are repeated the same number of times. A method for providing non-integer repetition is described in detail in co-pending U.S. Patent Application Serial No. 08/886,815, filed March 26, 1997, entitled "METHOD AND APPARATUS FOR TRANSMITTING HIGH SPEED DATA IN A SPREAD SPECTRUM COMMUNICATIONS SYSTEM", which is assigned to the assignee of the present invention and incorporated by reference herein.

The symbols from repetition generators **58a-58c** are provided to a corresponding one of interleavers **60a-60c** which reorders the repeated symbols in accordance with a predetermined interleaver format. Control processor **50** provides an interleaving format signal to each of interleavers **60a-60c** which indicates one of a predetermined set of interleaving formats. In the exemplary embodiment, the interleaving format is selected from a predetermined set of bit reversal interleaving formats.

The reordered symbols from interleavers **60a-60c** are provided to data scramblers **62a-62c**. Each of data scramblers **62a-62c** changes the sign of the data in accordance with a pseudonoise (PN) sequence. Each PN sequence is provided by passing a long PN code generated by long code or PN generator **82** at the chip rate through a decimator **84a-84c**, which selectively provides ones of the spreading symbols to provide a PN sequence at a rate no higher than that provided by PN generator **82**. Because the symbol rate on each carrier may be different from one another, the decimation rate of decimators **84a-84c** may be different. Decimators **84a-84c** are sample and hold circuits which sample the PN sequence out of PN generator **82** and continue to output that value for a predetermined period. The implementation of PN generator **82** and decimators **84a-84c** are well known in the art and are described in detail in the aforementioned U.S. Patent No. 5,103,459. Data scramblers **62a-62c** exclusively-OR the binary symbols from interleavers **60a-60c** with the decimated pseudonoise binary sequences from decimators **84a-84c**.

The binary scrambled symbol sequences are provided to serial to parallel converters (BINARY TO 4-LEVEL) **64a-64c**. Two binary symbols provided to converters **64a-64c** are mapped to a quaternary constellation with values ($\pm1, \pm1$). The constellation values are provided on two outputs from converters **62a-62c**. The symbol streams from converters **64a-64c** are separately provided to Walsh spreaders **66a-66c**.

There are many methods of providing high speed data in a code division multiple access communication system. In the preferred embodiment, the Walsh sequence length is varied in accordance with the rate of the data to be modulated. Shorter Walsh sequences are used to modulate higher speed data and longer Walsh sequences are used to modulate lower rate data. For example, a 64 bit Walsh sequence can be used to transmit data at 19.2 Ksps. However, a 32 bit Walsh sequence can be used to modulate data at 38.4 Ksps.

A system describing variable length Walsh sequence modulation is described in detail in co-pending U.S. Patent Application Serial No. 08/724,281, entitled "HIGH DATA RATE SUPPLEMENTAL CHANNEL FOR CDMA TELECOMMUNICATIONS SYSTEM", filed January 15, 1997 and incorporated by reference herein. The length of the Walsh sequences used to modulate the data depend on the rate of the rate of the data to be transmitted. FIG. 4 illustrates the Walsh functions in a traditional IS-95 CDMA system.

In the preferred embodiment of the invention, the number of Walsh channels allocated for the high-rate data can be any value 2^N where N = {2, 3, 4, 5, 6}. The Walsh codes used by Walsh coders **66a-66c** are $64/2^N$ symbols long, rather than the 64 symbols used with the IS-95 Walsh codes. In order for the high-rate channel to be orthogonal to the other code channels with 64-symbol Walsh codes, 2^N of the possible 64 quaternary-phase channels with 64-symbol Walsh are eliminated from use. Table I provides a list of the possible Walsh codes for each value of N and the corresponding sets of allocated 64-symbol Walsh codes.

N	$Walsh_i$	Allocated 64-Symbol Walsh Codes
2	+,+,+,+,+,+,+,+,+,+,+,+,+,+,+,+	0, 16, 32, 48
	+,−,+,−,+,−,+,−,+,−,+,−,+,−,+,−	1, 17, 33, 49
	+,+,−,−,+,+,−,−,+,+,−,−,+,+,−,−	2, 18, 34, 50
	+,−,−,+,+,−,−,+,+,−,−,+,+,−,−,+	3, 19, 35, 51
	+,+,+,+,−,−,−,−,+,+,+,+,−,−,−,−	4, 20, 36, 52
	+,−,+,−,−,+,−,+,+,−,+,−,−,+,−,+	5, 21, 37, 53
	+,+,−,−,−,−,+,+,+,+,−,−,−,−,+,+	6, 22, 38, 54
	+,−,−,+,−,+,+,−,+,−,−,+,−,+,+,−	7, 23, 39, 55
	+,+,+,+,+,+,+,+,−,−,−,−,−,−,−,−	8, 24, 40, 56
	+,−,+,−,+,−,+,−,−,+,−,+,−,+,−,+	9, 25, 41, 57
	+,+,−,−,+,+,−,−,−,−,+,+,−,−,+,+	10, 26, 42, 58
	+,−,−,+,+,−,−,+,−,+,+,−,−,+,+,−	11, 27, 43, 59
	+,+,+,+,−,−,−,−,−,−,−,−,+,+,+,+	12, 28, 44, 60
	+,−,+,−,−,+,−,+,−,+,−,+,+,−,+,−	13, 29, 45, 61
	+,+,−,−,−,−,+,+,−,−,+,+,+,+,−,−	14, 30, 46, 62
	+,−,−,+,−,+,+,−,−,+,+,−,+,−,−,+	15, 31, 47, 63
3	+,+,+,+,+,+,+,+	0, 8, 16, 24, 32, 40, 48, 56
	+,−,+,−,+,−,+,−	1, 9, 17, 25, 33, 41, 49, 57
	+,+,−,−,+,+,−,−	2, 10, 18, 26, 34, 42, 50, 58
	+,−,−,+,+,−,−,+	3, 11, 19, 27, 35, 43, 51, 59
	+,+,+,+,−,−,−,−	4, 12, 20, 28, 36, 44, 52, 60
	+,−,+,−,−,+,−,+	5, 13, 21, 29, 37, 45, 53, 61
	+,+,−,−,−,−,+,+	6, 14, 22, 30, 38, 46, 54, 62
	+,−,−,+,−,+,+,−	7, 15, 23, 31, 39, 47, 55, 63
4	+,+,+,+	0, 4, 8,…,60
	+,−,+,−	1, 5, 9, …,61
	+,+,−,−	2, 6, 10, …,62
	+,−,−,+	3, 7, 11, …,63
5	+,+	0, 2, 4,…,62
	+,−	1, 3, 5, …,63
6	+	0, 1, 2,…,63

Table I.

The + and – indicate a positive or negative integer value, where the preferred integer is 1. As is apparent, the number of Walsh symbols in each Walsh code varies as N varies, and in all instances is less than the number

of symbols in the IS-95 Walsh channel codes. Regardless of the length of the Walsh code, in the described embodiment of the invention the symbols are applied at a rate of 1.2288 Megachips per second (Mcps). Thus, shorter length Walsh codes are repeated more often. Control processor **50** provides a signal to Walsh coding elements **66a-66c** which indicates the Walsh sequence to be used to spread the data.

Alternative methods for transmitting high rate data in CDMA communication system also include methods generally referred to as channel bundling techniques. The present invention is equally applicable to the channel bundling methods for providing high speed data in a CDMA communciation system. One method of providing channel bundled data is to provide a plurality of Walsh channels for use by a signal user. This method is described in detail in the aforementioned U.S. Patent Application Serial No. 08/739,482. An alternative channel bundling technique is to provide the user with use of one Walsh code channel but to differentiate the signals from one another by means of different scrambling signals as described in detail in co-pending U.S. Patent Application Serial No. 08/838,240.

The Walsh spread data is provided to PN spreaders **68a-68c**, which apply a short PN sequence spreading on the output signals. In the exemplary embodiment, the PN spreading is performed by means of a complex multiplication as described in detail in the aforementioned co-pending U.S. Patent Application Serial No. 08/784,281. Data channels D_I and D_Q are complex multiplied, as the first real and imaginary terms respectively, with spreading codes PN_I and PN_Q, as the second real and imaginary terms respectively, yielding in-phase (or real) term X_I and quadrature-phase (or imaginary) term X_Q. Spreading codes PN_I and PN_Q are generated by spreading code generators **67** and **69**. Spreading codes PN_I and PN_Q are applied at 1.2288 Mcps. Equation (1) illustrates the complex multiplication performed.

$$(X_I + jX_Q) = (D_I + jD_Q)(PN_I + jPN_Q) \tag{1}$$

In-phase term X_I is then low-pass filtered to a 1.2288 MHz bandwidth (not shown) and upconverted by multiplication with in-phase carrier $COS(\omega_c t)$. Similarly, quadrature-phase term X_Q is low-pass filtered to a 1.2288 MHz bandwidth (not shown) and upconverted by multiplication with quadrature-phase carrier $SIN(\omega_c t)$. The upconverted X_I and X_Q terms are summed yielding forward link signal s(t). The complex multiplication

allows quadrature-phase channel set to remain orthogonal to the in-phase channel set and therefore to be provided without adding additional interference to the other channels transmitted over the same path with perfect receiver phase recovery.

The PN spread data is, then, provided to filters **70a-70c** which spectrally shape the signals for transmission. The filtered signals are provided to gain multipliers **72a-72c**, which amplify the signals for each carrier. The gain factor is supplied to gain elements **72a-72c** by control processor **50**. In the exemplary embodiment, control processor **50** selects the gain factor for each carrier in accordance with the channel condition and the rate of the information data to be transmitted on that carrier. As is known by one skilled in the art, data that is transmitted with repetition can be transmitted with lower symbol energy than data without repetition.

The amplified signals are provided to an optional switch **74**. Switch **74** provides the additional flexibility of channel hopping the data signals onto different carriers. Typically, switch **74** is only used when the number of carriers actually used to transmit the signal is smaller that the total number of possible carriers (3 in the present example).

The data is passed by switch **74** to carrier modulators **76a-76c.** Each of carrier modulators **76a-76c** upconvert the data to a different predetermined frequency. The upconverted signals are provided to transmitter **78** where they are combined with other similarly processed signals, filtered and amplified for transmission through antenna **80**. In the exemplary embodiment, the amplified frequency upon which each of the signals are transmitted varies with time. This provides additional frequency diversity for the transmitted signals. For example a signal that is currently being transmitted through carrier modulator **76a** will at predetermined time interval be switched so as to be transmitted on a different frequency through carrier modulators **76b** or **76c**. In accordance with a signal from control processor **50**, switch **74** directs an amplified input signal from gain multiplier **72a-72c** to an appropriate carrier modulator **76a-76c**.

Turning to FIG. 3, a receiver system embodying the present invention is illustrated. The signal received at antenna **100** is passed to receiver (RCVR) **102**, which amplifies and filters the signal before providing it to switch **104**. The data is provided through switch **104** to an appropriate carrier demodulator **106a-106c**. It will be understood by one skilled in the art that although the receiver structure is described for the reception of a signal transmitted on three frequencies, the present invention can easily be

extended to an arbitrary number of frequencies consecutive to one another or not.

When the carriers on which the data is transmitted are rotated or hopped to provide additional frequency diversity, switch **104** provides the received signal to a selected carrier demodulator **106a-106c** in response to a control signal from control processor **125**. When the carrier frequencies are not hopped or rotated, then switch **104** is unnecessary. Each of carrier demodulators **106a-106c** Quaternary Phase Shift Keying (QPSK) demodulate the received signal to baseband using a different downconversion frequency to provide a separate I and Q baseband signals.

The downconverted signals from each of carrier demodulators **106a-106c** are provided to a corresponding PN despreader **108a-108c** which removes the short code spreading from the downconverted data. The I and Q signals are despread by complex multiplication with a pair of short PN code. The PN despread data is provided to Walsh demodulators **110a-110c**, which uncover the data in accordance with the assigned code channel sequences. In the exemplary embodiment, Walsh functions are used in the generation and reception of the CDMA signals but other forms of code channel generation are equally applicable. Control processor **125** provides a signal to Walsh demodulators **110a-110c** indicating the Walsh sequences to be used to uncover the data.

The Walsh despread symbols are provided to parallel-to-serial converters (4-LEVEL TO BINARY) **112a-112c**, which map the 2-dimensional signal into a 1-dimensional signal. The symbols are then provided to descramblers **114a-114c**. Descramblers **114a-114c** descramble the data in accordance with a decimated long code sequence generated as described with respect to the decimated long code sequences used to scramble the data in FIG. 2.

The descrambled data is provided to de-interleavers (DE-INT) **116a-116c**. De-interleavers **116a-116c** reorder the symbols in accordance with selected de-interleaver formats that are provided by control processor **125**. In the exemplary embodiment, control processor **125** provides a signal indicative of the size of the deinterleaver and the scheme of de-interleaving to each of de-interleavers **116a-116c**. In the exemplary embodiment, the de-interleaving scheme is selected from a predetermined set of bit reversal de-interleaving schemes.

The de-interleaved symbols are then provided to symbol combiners **118a-118c** which coherently combine those repeatedly transmitted symbols. The combined symbols (soft decisions) are then provided to variable ratio

multiplexer **120** which reassembles the data stream and provides the reassembled data stream to decoder **122**. In the exemplary embodiment decoder **122** is a maximum likelihood decoder, the implementation of which is well known in the art. In the exemplary embodiment, decoder **122** contains a buffer (not shown) which waits until an entire frame of data has been provided to it before beginning the decoding process. The decoded frame is provided to CRC check means **124** which determines whether the CRC bits check and if so provides them to the user otherwise an erasure is declared.

Having thus described the invention by reference to a preferred embodiment it is to be well understood that the embodiment in question is exemplary only and that modifications and variations such as will occur to those possessed of appropriate knowledge and skills may be made without departure from the spirit and scope of the invention as set forth in the appended claims and equivalents thereof.

I CLAIM:

CLAIMS

1. A transmitter for transmitting data at a data rate in a plurality of channels each having a capacity less than the data rate, the transmitter comprising;

a controller for determining the capacity of each of a plurality channels and selecting a data rate for each channel depending on the determined capacity;

a plurality of transmission subsystems responsive to the controller and each associated with a respective one of the plurality of channels for scrambling encoded data with codes unique to the channel for transmission in the channel; and

a variable demultilplexer responsive to the controller for demultiplexing the encoded data into the plurality of transmission subsystems at a demultiplexing rate derived from the data rates selected for the channels by the controller.

2. A transmitter as claimed in claim 1, further comprising an encoder for generating the encoded data from frames of data input thereto.

3. A transmitter as claimed in claim 1 or 2, wherein each transmission subsystem comprises a symbol repetition unit for repeating symbols to output the same at a rate corresponding to the rate selected for the channel by the controller.

4. A transmitter as claimed in claim 3, wherein each transmission subsystem comprises an interleaving unit for reordering the repeated symbols depending on an interleaving format determined by the controller.

5. A transmitter as claimed in claim 4, further comprising a long code generator for generating a respective long code for each channel; and

in each transmission subsystem, a scrambler for scrambling the reordered symbols using the respective code for the channel.

6. A transmitter as claimed in claim 5, wherein the long code generator comprises for each channel a decimator unit for decimating a generated long code at a decimation rate determined by the controller so as to produce the respective long codes for each channel.

7. A transmitter as claimed in claim 6, further comprising variable coding units in each transmission subsystem for modulating the scrambled symbols from the scrambler.

8. A transmitter as claimed in claim 7, wherein the coding units are arranged to modulate the scrambled symbols with a respective walsh code.

9. A transmitter as claimed in claim 7 or 8, further comprising a pseudo noise spreader in each channel for spreading the modulated symbols.

10. A transmitter as claimed in any preceding claim, further comprising:

a switch; and

a plurality of carrier modulators, wherein the switch is responsive to the controller for switching the scrambled data from the plurality of transmission subsystems between the plural carrier modulators for modulation of the signals onto different carriers at different times.

11. A receiver comprising:

a receiving circuit for receiving signals simultaneously in a plurality of channels each of which signals define scrambled encoded symbols which together represent data from a common origin;

a controller for determining a symbol rate for the signals in each channel;

a plurality of receiving subsystems responsive to the controller and each associated with a respective one of the plurality of channels for descrambling encoded symbols with codes unique to the channel to enable the data to be extracted therefrom; and

a variable multilplexer responsive to the controller for multiplexing the data from the plurality of receiving subsystems at a multiplexing rate derived from the symbol rates determined for the channels by the controller onto an output.

12. A receiver as claimed in claim 11, further comprising an decoder for decoding the encoded data output from the multiplexer into frames of data.

13. A receiver as claimed in claim 11 or 12, further comprising a pseudo noise despreader in each channel for despreading the scrambled encoded symbols.

14. A receiver as claimed in claim 13, further comprising variable decoding units in each receiving subsystem for demodulating the despread symbols from the despreader.

15. A receiver as claimed in claim 14, wherein the decoding units are arranged to demodulate the despread symbols with a respective walsh code.

16. A receiver as claimed in claim 15, further comprising, in each receiving subsystem, a descrambler for descrambling the despread symbols using a respective long code for the channel.

17. A receiver as claimed in claim 16, wherein each receiving subsystem comprises an deinterleaving unit for reordering the repeated symbols depending on an interleaving format determined by the controller.

18. A receiver as claimed in claim 17, wherein each receiving subsystem comprises a symbol combiner for combining symbols to output the same to the demultiplexer at a rate corresponding to the rate determined for the channel by the controller.

19. A receiver as claimed in any of claims 11 to 18, further comprising:

a switch; and

a plurality of carrier demodulators, wherein the switch is responsive to the controller for switching the received signals between the plural carrier demodulators for demodulation of the signals into different receiving subsystems at different times.

20. A wireless transmitter, comprising:

encoder for receive a set of information bits and encoding said information bits to provide a set of code symbols; and

transmission subsystem for receiving said code symbols and for providing a subset of said code symbols on a first carrier frequency and the remaining symbols on at least one additional carrier frequency.

21. A method of transmitting data at a data rate in a plurality of channels each having a capacity less than the data rate, the method comprising;

determining the capacity of each of a plurality channels and selecting a data rate for each channel depending on the determined capacity;

scrambling encoded data with codes unique to the channel for transmission in the channel; and

demultiplexing the encoded data into the plurality of channels at a demultiplexing rate derived from the data rates selected for the channels by the controller.

22. A method as claimed in claim 21, further comprising an encoder for generating the encoded data from frames of data input thereto.

23. A method as claimed in claim 21 or 22, further comprising repeating symbols for each channel to output the same at a rate corresponding to the rate selected for the channel.

24. A method as claimed in claim 23, further comprising reordering the repeated symbols depending on a determined interleaving format.

25. A method as claimed in claim 24, further comprising

generating a respective long code for each channel; and

scrambling the reordered symbols in each transmission subsystem using the respective code for the channel.

26. A method as claimed in claim 25, wherein the long code is generated for each by decimating a generated long code at a determined decimation rate for each channel.

27. A method as claimed in claim 26, further comprising for modulating the scrambled symbols with a code.

28. A method as claimed in claim 27, the scrambled symbols are modulated with a respective walsh code.

29. A method as claimed in claim 27 or 28, further comprising spreading the modulated symbols with pseudo noise.

30. A method as claimed in any of claims 21 to 29, further comprising modulating the scrambled data onto different carriers at different times.

31. A method of receiving data, the method comprising:

receiving signals simultaneously in a plurality of channels each of which signals define scrambled encoded symbols which together represent data from a common origin;

determining a symbol rate for the signals in each channel;

descrambling encoded symbols in each channel with codes unique to the channel to enable the data to be extracted therefrom; and

multiplexing the descrambled data from the plurality of channels at a multiplexing rate derived from the symbol rates determined for the channels.

32. A method of receiving data as claimed in claim 31, further comprising decoding the multiplexed encoded data into frames of data.

33. A method of receiving data as claimed in claim 31 or 32, further comprising despreading the scrambled encoded symbols using a pseudo noise code.

34. A method of receiving data as claimed in claim 33, further comprising demodulating the despread symbols by way of variable decoding.

35. A method of receiving data as claimed in claim 34, wherein the despread symbols are demodulated with a respective walsh code.

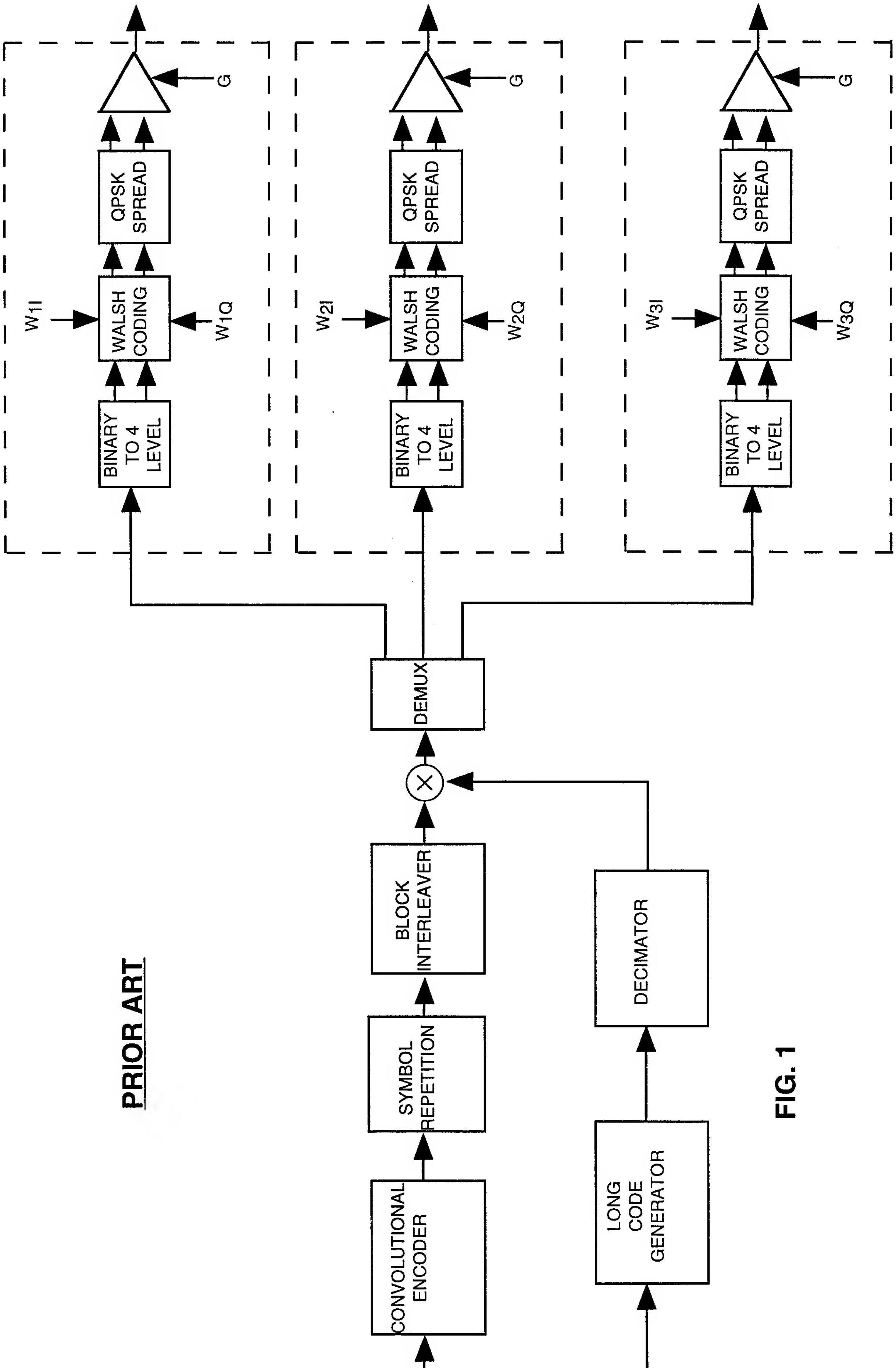
36. A method of receiving data as claimed in claim 35, further comprising descrambling the despread symbols in each channel using a respective long code for the channel.

37. A method of receiving data as claimed in claim 36, further comprising reordering the repeated symbols depending on a determined interleaving format.

38. A method of receiving data as claimed in claim 37, further comprising combining symbols in a channel before demultiplexing the same at a rate corresponding to the rate determined for the channel.

39. A method of receiving data as claimed in any of claims 31 to 38, further comprising:

demodulating the signals in different channels at different times.



PRIOR ART

FIG. 1

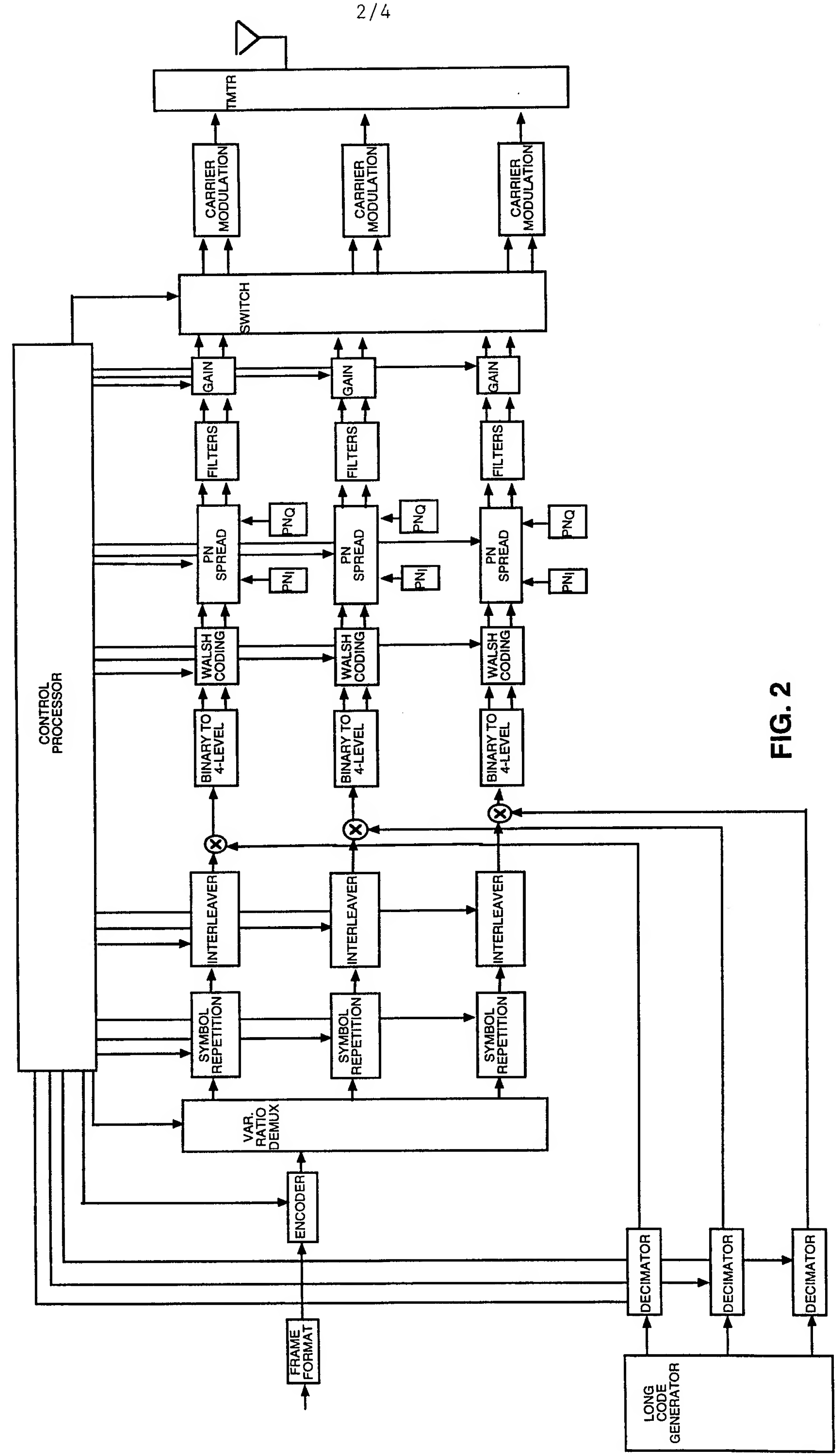

TMTR
CARRIER MODULATION
CARRIER MODULATION
CARRIER MODULATION
SWITCH
CONTROL PROCESSOR
GAIN
GAIN
GAIN
FILTERS
FILTERS
FILTERS
PN SPREAD
PN SPREAD
PN SPREAD
PNQ
PNQ
PNQ
PNI
PNI
PNI
WALSH CODING
WALSH CODING
WALSH CODING
BINARY TO 4-LEVEL
BINARY TO 4-LEVEL
BINARY TO 4-LEVEL
INTERLEAVER
INTERLEAVER
INTERLEAVER
SYMBOL REPETITION
SYMBOL REPETITION
SYMBOL REPETITION
VAR. RATIO DEMUX
ENCODER
FRAME FORMAT
DECIMATOR
DECIMATOR
DECIMATOR
LONG CODE GENERATOR

FIG. 2

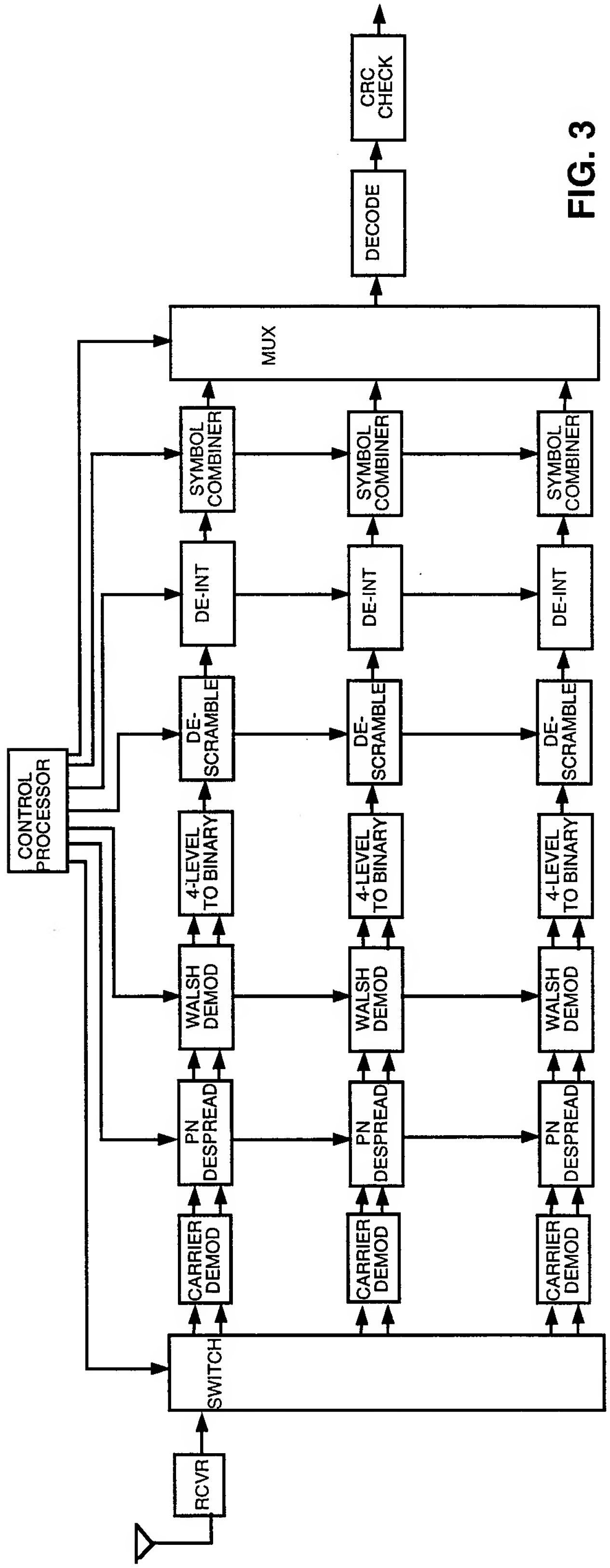

RCVR
SWITCH
CARRIER DEMOD
PN DESPREAD
WALSH DEMOD
4-LEVEL TO BINARY
DE-SCRAMBLE
DE-INT
SYMBOL COMBINER
CONTROL PROCESSOR
MUX
DECODE
CRC CHECK

FIG. 3

Walsh Chip within a Walsh Function

Walsh Function Index	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31	32	33	34	35	36	37	38	39	40	41	42	43	44	45	46	47	48	49	50	51	52	53	54	55	56	57	58	59	60	61	62	63
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
2	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1
3	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
4	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1
5	0	1	0	1	1	0	1	0	0	1	0	1	1	0	1	0	0	1	0	1	1	0	1	0	0	1	0	1	1	0	1	0	0	1	0	1	1	0	1	0	0	1	0	1	1	0	1	0	0	1	0	1	1	0	1	0	0	1	0	1	1	0	1	0
6	0	0	1	1	1	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1	0	0	0	0	1	1	1	1	0	0
7	0	1	1	0	1	0	0	1	0	1	1	0	1	0	0	1	0	1	1	0	1	0	0	1	0	1	1	0	1	0	0	1	0	1	1	0	1	0	0	1	0	1	1	0	1	0	0	1	0	1	1	0	1	0	0	1	0	1	1	0	1	0	0	1
8	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1
9	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0
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11	0	1	1	0	0	1	1	0	1	0	0	1	1	0	0	1	0	1	1	0	0	1	1	0	1	0	0	1	1	0	0	1	0	1	1	0	0	1	1	0	1	0	0	1	1	0	0	1	0	1	1	0	0	1	1	0	1	0	0	1	1	0	0	1
12	0	0	0	0	1	1	1	1	1	1	1	1	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	0	0	0	0
13	0	1	0	1	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	0	1	0	1
14	0	0	1	1	1	1	0	0	1	1	0	0	0	0	1	1	0	0	1	1	1	1	0	0	1	1	0	0	0	0	1	1	0	0	1	1	1	1	0	0	1	1	0	0	0	0	1	1	0	0	1	1	1	1	0	0	1	1	0	0	0	0	1	1
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17	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0
18	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0
19	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1
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24	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	1	0	0	0	0	0	0	0	0
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57	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0
58	0	0	1	1	0	0	1	1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0	0	0	1	1	0	0	1	1	1	1	0	0	1	1	0	0	0	0	1	1	0	0	1	1	0	0	1	1	0	0	1	1	1	1	0	0	1	1	0	0
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61	0	1	0	1	1	0	1	0	1	0	1	0	0	1	0	1	1	0	1	0	0	1	0	1	0	1	0	1	1	0	1	0	1	0	1	0	0	1	0	1	0	1	0	1	1	0	1	0	0	1	0	1	1	0	1	0	1	0	1	0	0	1	0	1
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FIG. 4